Signal Processing for Smart Sensor Arrays: From Research to Application-Rich Technology Insertion

Workshop Proceedings

Arlington, Virginia
27–28 April 1995

Edited by
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Supported by
The National Science Foundation
Circuits and Signal Processing Program
Microelectronics Information Processing Systems Division
Computer and Information Science and Engineering Directorate

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In April 1995, a two-day workshop/panel on Array Signal Processing (ASP) was held at the National Science Foundation. Presentations were made by Principal Investigators supported by the Circuits and Signal Processing Program of the MIPS Division of NSF and by selected researchers from industry. Program directors from different funding agencies attended the workshop and participated in the panel discussions. Each speaker (i) summarized their respective research program in terms of goals, potential impact, progress to date, milestones and/or setbacks, (2) discussed the major research/technological problems facing their respective area of ASP, and (3) provided an overall assessment of the field including recommendations for future action. The program for this workshop is listed on the next two pages.

Each presenter at the workshop contributed to this document a concise summary of their presentation, including illustrative figures and flowcharts, performance curves, and, in some cases, photos of prototype experimental array systems. These summaries constitute the proceedings of the workshop in Chapter 3. Specific application areas addressed included wireless/mobile communications, sound reinforcement, ultrasonic imaging, medical imaging, GPS based navigation, rapid thermal processing for semiconductor manufacturing, and synthetic aperture radar. A consensus on findings and recommendations for the overall field of ASP based on panel discussions is presented in Chapters 1 and 2.

Program

THURSDAY AM, 27 APRIL 1995

I. SMART ANTENNA ARRAYS FOR MOBILE COMMUNICATIONS

8:30 **Blind Adaptive Beamforming for Digital Cellular/PCS Base Station Antenna Arrays**
Michael Zoltowski, Purdue University

8:50 **Summary of Smart Antenna Research at UT-Austin**
Ganghuan Xu, University of Texas at Austin

9:10 **Issues and Challenges in Array Signal Processing for Communications Applications**
Lee Swindlehurst, Brigham Young University

9:30 **Theoretical Results on CDMA Detection Using Antenna Arrays**
Mos Kaveh, University of Minnesota

9:50 **PANEL SESSION**

II. ACOUSTIC ARRAY PROCESSING

10:40 **Processing of Microphone Arrays for Spatially-Selective Sound Capture**
James Flanagan, Rutgers University

11:00 **Calibration and Experiments For Signal-Subspace Detection & Estimation Using an Ultrasound-in-Air Array**
Mos Kaveh, University of Minnesota

11:20 **Adaptive Microphone Arrays for Tracking and Beamforming**
Eric Dowling, University of Texas at Dallas

11:40 **PANEL SESSION**

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III. DEVELOPING TIES WITH INDUSTRY

1:30 **DIRECTED DISCUSSION**

IV. ARRAY DESIGN ISSUES

2:20 **Redundancy in Active Imaging Arrays**
Saleem Kassam, University of Pennsylvania

2:40 **Distributed Detection With Incomplete Knowledge**
Rick Blum, Lehigh University

3:00 **Maximizing the Accuracy and Resolution Capacity of Sensor Array Direction Finding**
Yoram Bresler, University of Illinois

3:20 **PANEL SESSION**

V. SIGNAL SUBSPACE ALGORITHMS FOR DETECTION & PARAMETER ESTIMATION

4:10 **Utilizing Sensor and Wave Properties for Antenna Array Processing**
Jian Li, University of Florida

4:30 **Subspace Applications in Array Processing**
Barry Van Veen, University of Wisconsin

4:50 **Worst-Case Bounds and Globally Convergent ML Algorithms For Parameter Estimation of Superimposed Signals**
Yoram Bresler, University of Illinois

5:10 **PANEL SESSION**

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VI. TOWARDS REAL-TIME IMPLEMENTATION OF SIGNAL SUBSPACE ALGORITHMS

8:30 **Sphericalized Subspace Updating**
Ron DeGroat, University of Texas at Dallas

8:50 **Some Results in Algorithms and Architectures Development**
Ray Liu, University of Maryland

9:10 **PANEL SESSION**

VII. EXPERIMENTAL ARRAY SYSTEMS

10:10 **Experimental Sensor Array Systems for Mobile Communications and Semiconductor Manufacturing**
Ganghuan Xu, University of Texas at Austin

10:30 **An Experimental Program in RF Environment Characterization for SDMA-Based Wireless Communication Systems**
Richard Roy, ArrayCom

10:50 **Adaptive Arrays for Wireless Communication Systems With Multipath**
Jack Winters, AT&T Bell Laboratories

11:10 **Tradeoffs Among Hardware, Software and Algorithms so That a "Small" Group Can Build a Large Array**
Harvey Silverman, Brown University

11:30 **PANEL SESSION**

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VIII. "ARRAY" OF APPLICATIONS

1:30 **Novel Algorithms for Finding Localized Energy Solutions With Application to Magnetoencephalography (MEG)**
Bhaskar Rao, University of California at San Diego

1:50 **Biomedical Applications of Array Signal Processing**
Kevin Buckley, University of Minnesota

2:10 **SAR Image Formation and Processing**
Jian Li, University of Florida

2:30 **Null Steering/ Beamforming Arrays Used in Conjunction With GPS Receivers**
Anton S. Gecan, E-Systems

2:50 **PANEL SESSION**

IX. EXECUTIVE SUMMARY

3:30 **DIRECTED DISCUSSION**

Contents

Chapter 1

Executive Summary

1.1 The Workshop

In April 1995, a two-day workshop/panel on Array Signal Processing (ASP) was held at the National Science Foundation. This report summarizes the proceedings of that meeting and of email conversations before and after the meeting. The participants included active researchers from academia, industry, and government in a variety of fields (see Appendices A & B).

This workshop and the attendant panel discussion was modeled after the *Signal Processing for the National Information Infrasructure (NII) Workshop/Panel* convened by the National Science Foundation during 18-19 August 1994 and chaired by Professor Robert M. Gray of Stanford University. The format of the Executive Summary portion of this document follows closely that of the Workshop/Panel Report compiled by Professor Gray.

1.2 Overview

Array Signal Processing (ASP) deals with the multidimensional processing of the signals received at a collection of sensors spaced at discrete points over some aperture in a multicomponent signal wavefield. One primary attribute of ASP is the enormous breadth of applications. The specific application areas discussed in depth at the workshop are listed below.

- wireless/mobile communications
- sound reinforcement
- ultrasonic imaging
- medical imaging
- GPS (Global Positioning System) based navigation
- rapid thermal processing for semiconductor manufacturing
- synthetic aperture radar
- radio direction finding

The primary focus of the workshop was on so-called smart sensor arrays that automatically adapt to their environment in combining the outputs of the individual sensors to strengthen the desired signal while simultaneously suppressing interference and noise.

1.2.1 Smart Antenna Arrays for Mobile/Wireless Communications

As a result of the recent and ongoing explosive growth of the wireless communications industry, there has been a major resurgence of interest in the use of smart antenna arrays as a means to improve both the channel capacity and the quality of wireless communications – cellular telephony, personal communications services (PCS), and wireless local area networks. Thus, the use of smart antenna arrays for wireless communications received much attention at the workshop.

Sensor array processing for wireless communications may be viewed as "the new frontier." Many years of research and development have gone into the design and implementation of currently deployed wireless systems, and their commercial impact is unquestionably being felt worldwide. These systems, however, face fundamental capacity limitations that promise to slow their penetration substantially once the "squeeze" is felt. Efficient exploitation of recently developed multidimensional signal processing tools for ASP has the potential of lifting these systems to the "next level," providing for a significant increase in products and services for the wireless consumer, expanding markets for more sophisticated products from manufacturers, and ultimately providing for increased productivity and jobs in the private telecommunications economic sector.

1.3 Findings and Recommendations

The findings and recommendations highlighted below were developed over two days of active discussion at the workshop and through extensive email exchanges in the weeks surrounding the event. Chapter 2 expands these findings.

1.3.1 Grand Challenges

Establish testbed(s) for real-time experimental validation

> Live tests demonstrating an easily perceived improvement in voice quality in nonstationary, multipath rich environments should be a high priority goal. This entails either gaining access to or building experimental antenna arrays with real-time processing capabilities. Specifically, this entails the development of facilities that enable the user to implement (via DSP's and /or FPGA's) and measure the performance of the algorithms under consideration. Similarly, relative to smart antenna arrays for PCS, live tests showing a significant measurable increase in data throughput should be a high priority goal.

Develop (better) methods to assess the performance of algorithms on real data

> The problem of ascertaining reliable information on ground truth when dealing with the processing of real data is of paramount importance, especially in nonstationary, multipath rich environments. New concepts are sorely needed to deal with this all-pervasive, vexing issue. There is also a need to develop acceptable and trustworthy protocols for validating the utility and quality of angle-of-arrival estimates, time delay estimates, frequency estimates, etc., obtained by ASP algorithms, and to develop the means of predicting utility and quality from easily made quantitative measurements.

As Array Signal Processing is a sub-discipline of the overall field of Signal Processing, a number of the findings and recommendations arrived at by the aforementioned *Signal Processing for the NII Workshop Panel* and succinctly articulated by Professor Robert M. Gray in the attendant Workshop/Panel Report apply here, *mutatis mutandis.* Thus, except for minor editing changes, several of the findings and recommendations listed below were copied verbatim from the *Signal Processing for the NII Workshop/Panel Report.*

Industrial Collaboration

Finding:
Theory and algorithms for ASP must be developed in the context of intended applications and in close cooperation with users if the algorithms are to reflect the needs of the user community. This involves close collaboration with industry, and possibly national and government laboratories as well.
Recommendation:
Genuine cooperative efforts for research on smart sensor arrays amongst academia, government and industry should be initiated and supported to the greatest extent possible.

Standard Databases

Finding:
Theoretical development of ASP algorithms can often take place using simulated data for preliminary tests, but genuine comparisons among competing algorithms and validation of quality and utility require generally available standard data sets obtained from experimental array systems sanctioned by the ASP community. Hence, common standard databases enhance research and permit comparison between competing schemes.
Recommendation:
Support the design and establishment of standard databases for ASP research. This can be done either through financial support to individual institutions (as may be necessary for expensive databases) or by providing a clearing house of publicly available databases, reachable via the NII. For example, these databases

could be stored at the signal processing database at Rice University or on the World Wide Web.

Standards for Algorithm Validation

Finding: It will be useful to develop standards to validate new ASP algorithms, that is, to demonstrate quantitatively that they perform as claimed or intended.
Recommendation:
Support the development of standards for validation of ASP algorithms. This will be strongly application dependent, but good ideas will benefit several applications.

Public Education about ASP

Finding:
Commercial companies, including small start-up companies as well as large corporations, need to know what smart sensor arrays can do for them, and they don't necessarily read the *IEEE Transactions* or attend *IEEE* sponsored conferences.
Recommendation:
Fund exploratory R&D targeted at bringing ASP concepts to the general public. In addition, each PI should be strongly encouraged to inform the public about the commercial impact of their research.

1.4 Conclusions

The primary goal of this report is to inform both research agencies and the general SP community of the key role of ASP in future commercial products and to identify important trends in the development of ASP for commercial applications. We believe that active cooperation among the SP community and many of the user communities will provide the fastest possible development of new and widely useful ASP algorithms. The National Science Foundation and other Government agencies can best assist such development by encouraging the necessary cooperative research and development efforts by whatever means possible. The balance of support should be tilted toward the very best research rather than incremental improvements of mature methods or unsubstantiated claims of allegedly novel methods.

Chapter 2

Overview of Array Signal Processing and Recommendations

2.1 Introduction

Array Signal Processing deals with the processing of signals that arise from multiple sources, typically from an array or collection of sensors. The spectrum of applications of ASP is extremely broad as evidenced by the specific application areas discussed in depth at the workshop listed in Section 1.2. Other major application areas of ASP not addressed in detail at the workshop include sonar, exploration seismology, radio astronomy, and electromagnetic hyperthermia treatment of cancer.

The type of sensor employed depends on the application. In wireless communications, the sensors are antennas which convert electromagnetic energy into electrical signals. In sound reinforcement, the sensors are microphones which convert variations in air pressure to electrical signals. In sonar, variations in water pressure are converted similarly through the use of hydrophones. In radio astronomy, the sensors are large parabolic reflectors such as those comprising the VLA (Very Large Array) in New Mexico which probes the universe. In the case of Computed Tomography, an important medical imaging modality that generates CAT scans, for example, the sensors are X-ray detectors.

2.1.1 Smart Antenna Arrays for Mobile/Wireless Communications

Adaptive antenna arrays is a classical example of a dual use technology. For many years, adaptive antenna arrays were primarily used in military applications for enhanced LPI (Low Probability of Intercept) through electronic beamforming effecting high directionality and null steering in the transmit mode, RF (Radio Frequency) source localization (for Electronic Signal Warfare) via high-resolution direction finding, and jammer cancellation via null steering in the receive mode.

The traditional purpose of antenna arrays in commercial wireless communications is to provide spatial diversity for countering fading due to multipath. Recently, there has been a resurgence of interest in the use of antenna arrays as a means to improve both the channel capacity and quality of wireless communications – cellular telephony, personal communications services (PCS), and wireless local area networks – – by exploiting the enhanced interference rejection capability they provide. Co-channel signals may be discriminated based on their angle-of-arrival at the antenna array site, as well as on their temporal and/or spectral characteristics.

The world market for base station antennas for digital wireless applications is projected to grow at an unprecedented rate. Cellular carriers in the US made need 15,000 new cell sites over the next decade, to upgrade their services and meet the anticipated demand. PCS services may require an additional 100,000 sites. According to a report in *Microwaves & RF* (May, 1995), "growth for this segment of the antenna market will increase at a rate greater than the total worldwide computer market during at least the next five years." It is expected to climb from \$64.2 million, in 1994, to \$325.5 million, in the year 2000. The trend is toward higher frequencies (above 900 MHz), lower-power levels, more-compact designs, and adaptive features.

2.1.2 Recent Evolution of ASP Research

Approximately fifteen years ago, ASP schemes found new prominence with the advent of so-called eigenstructure-based or signal subspace methods. The premise of these techniques is parametric modeling of the signals of interest. Incorporating a parametric model into the attendant algorithm allows one to achieve higher resolution than nonparametric techniques, such as those based simply on Fourier analysis, for example.

The fundamental principle underlying the eigenstructure based or signal subspace methods is that each point source in the field of the view of the sensor array makes a rank one contribution to the spatial correlation matrix of array outputs. The class of signal subspace algorithms based on the MUSIC algorithm - for MUltiple SIgnal Classification - developed by Ralph Schmidt at Stanford University in 1979, extract directional parameters about the incident wavefronts by exploiting the orthogonality between the array manifold vectors associated with the incident wavefronts, one per wavefront, and each of the noise eigenvectors of the spatial correlation matrix, i.e., those eigenvectors associated with the smallest eigenvalues. The array manifold for a given source is a vector that describes the relative phase distribution induced across the array aperture due to the signal arriving from that source.

The class of signal subspace algorithms based on the ESPRIT algorithm - for Estimation of Signal Parameters by Rotational Invariance Techniques - developed by Paulraj, Roy, and Kailath at Stanford University in 1985, utilizes a special invariance structure present in certain array geometries to extract the directional parameters of the incident wavefronts from the generalized eigenvalues of submatrices of the spatial correlation matrix. Compared to MUSIC it overcomes the need for array calibration, improves the estimation accuracy, and reduces the computational requirements.

However, despite the many exciting advances in ASP realized during the last decade, many of the old, unresolved stumbling blocks are still there, waiting to "clothesline" the unsuspecting, e.g., array calibration, model order estimation, model mismatch, etc. Efforts must be made to exploit any known temporal characteristics of the signals of interest to make the spatial processing as robust as possible. For example, digital communications signals have a rich structure that should be exploited to facilitate robust space-time (adaptive) processing. Thus, there are a host of challenging, albeit, important problems awaiting solution, the resolution of which will unquestionably accelerate the adoption of these tools in the wireless arena.

2.2 Recommendations

Grand Challenges

2.2.1 Establish testbed(s) for real-time experimental validation

PI's need to spend more time demonstrating the efficacy of ASP in a truly convincing manner. In particular, not just processing several frames of (pseudo-)synthetic data, and claiming superiority over competing approaches if the direction(s) of arrival estimates appear to be closer to what was (naively) anticipated, or some equivalent; rather, exercising the algorithm "on line," and evaluating performance by more realistic means (e.g., perceptual measures).

Live tests demonstrating a easily perceived improvement in voice quality in nonstationary, multipath rich environments should be a high priority goal, the ultimate goal. This entails either gaining access to or building experimental antenna arrays with real-time processing capabilities. Specifically, facilities that enable the user to implement (via DSP's and /or FPGA's) and measure the performance of the algorithms under consideration. Similarly, relative to smart antenna arrays for PCS, live tests showing a significant measurable increase in data throughput should be a high priority goal.

The following comments from one of the participants (Professor Rick Blum of Lehigh University) are relevant:

> "I have worked in industry for seven years and part of this time was spent as an analog and digital electronics designer (for communications). Some present at this PI meeting continually emphasized the difficulty of implementing theoretical algorithms in "real" systems. While I agree that implementing theoretical algorithms can be challenging, the difficulties can typically be overcome with a reasonable amount of effort. For this reason, I feel that the difficulty of implementing theoretical algorithms was somewhat overstated."

2.2.2 Develop (better) methods to assess the performance of algorithms on real data

The problem of ascertaining reliable information on ground truth when dealing with the processing of real data is of paramount importance, especially in nonstationary, multipath rich environments. New concepts are sorely needed to deal with this all-pervasive, vexing issue. We also need to develop acceptable and trustworthy protocols for validating the utility and quality of angle-of-arrival estimates, time delay estimates, phase estimates, etc., obtained by ASP algorithms, and to develop the means of predicting utility and quality from easily made quantitative measurements.

2.2.3 Industrial Collaboration

PI's need to develop closer ties to industry. ASP research goals can be identified that are common to a variety of applications, but the attendant theory and algorithms must be developed in the context of the intended applications if the algorithms are to truly reflect the needs of the user community while having the highest possible efficiency and performance. This generally requires close interaction with industry, possibly involving time spent (physically) by the PI at an industrial site.

Each PI should work towards giving several talks at interested companies on their ASP research each year. Engineers in industry do not, in general, have the time or background to read the *IEEE Transactions*, or even to attend some of the relevant conferences. Most companies have some kind of formal or informal seminar series and are very open to hearing first hand relevant university research. In addition, industry can provide feedback to make the research possibly more relevant and also introduce the PI to practical problems that he/she was not aware of but has the expertise to tackle.

The burden is on PIs to demonstrate to industry that they provide a service that can help industry increase their profits. In today's environment this is much more important than in the past. The government can also help by giving incentives to encourage industry to support university research. Once a company supports some research they will usually interact much more with the investigator they are supporting.

Genuine cooperative efforts for research on ASP among academia, government, and industry should be initiated by every party and supported to the greatest extent possible. The NSF program GOALI is just one means of several by which the development of ASP algorithms can be encouraged within the context of specific applications through collaborative research and development.

2.2.4 Standard Databases

Theoretical development of ASP algorithms can often take place using simulated data for preliminary tests, but genuine comparisons among competing algorithms and validation of quality and utility require generally available standard data sets. Hence common standard databases can enhance research and ease comparison among existing schemes and competing new schemes.

A team of investigators should carefully design a set of scenarios for which data will be collected, and then collect this data from appropriate testbeds. Industry and university experts should define the "scenarios of interest" to insure that these scenarios are realistic and to encourage industry-university interaction (see below).The resulting data sets could either be placed on the signal processing database at Rice University or on the World Wide Web. The existence of these databases and instructions for their use should be made widely available to relevant SP newsgroups and professional newsletters.

Along with the data should be substantial documentation of the experimental array system that generated the data, including calibration procedures employed as well as overall system parameters and some specifics on the components comprising the array. Principal Investigators who build experimental array systems with NSF Equipment Grant money, or through various other means of NSF support, should be strongly encouraged to generate data for public access via the Internet through some one of the options described above.

2.2.5 Standards for Algorithm Validation

Not all new SP algorithms developed over time can or should be made readily available for use by the general array signal processing community, which includes researchers and engineers in industry, government laboratories, national laboratories, and academia. De facto standards exist simply because they work well and no clearly superior algorithms performing similar or better functions have appeared. Nonetheless, it will be useful to develop standards to validate new algorithms, that is, to demonstrate quantitatively that they perform as claimed or intended. Thus, the development of standards for validation of ASP tools should be supported. This will be strongly application dependent, but good ideas will benefit several applications.

2.2.6 Public Education about ASP

Efforts should be made to teach senior EE undergraduates in communications and/or signal processing courses about array signal processing (e.g., smart antenna arrays for wireless communications.) Array signal processing needs to become a mainstream item covered in significant depth at the Master's level.

Chapter 3

Proceedings of the Workshop

3.1 Smart Antenna Arrays for Mobile Wireless Communications

3.1.1 Issues and Challenges in Array Signal Processing for Communications Applications

Lee Swindlehurst, Brigham Young University

Background

The principal driving force behind the field of array signal processing was originally military surveillance applications, typically characterized by the following:

- Non-cooperative emitters
- Often little known about signal environment
- Interested in emitter location as well as signal interception
- One-of-a-kind systems

Interest is now growing in the use of array signal processing techniques in commercial communications systems, where one may assume

- Cooperative transmission
- Known signal structure
- Less interested in emitter location
- System cost & complexity very important

Consequently, there have been fundamental changes in the goals of research in this area.

One prominent communications application for array signal processing is in the field of mobile outdoor radio, where it is envisioned that "smart" (adaptive) antenna arrays at cellular basestations can be used to increase transmission range and capacity. Figure 3.1 depicts a typical mobile radio signal environment, characterized by

- Relatively slow data rates (tens of Kbps)
- Large multipath delay and amplitude spread possible due to local scattering, urban high rises, mountains, etc.
- Dynamic multipath (remotes moving at 60 mph, 150 Hz fade rates)
- Unknown co-channel interference, either from within the cell or from adjacent cells
- Split frequency communication, remote and basestation communicate over several frequency/time channels

Indoor Personal Communications Systems (PCS) are another arena where adaptive arrays have been considered, particularly by Bell Labs. The indoor PCS environment differs from the outdoor mobile radio environment in the following ways:

- Faster data rates (tens to hundreds of Mbps)
- Less delay and amplitude spread
- Slower multipath variation (tens of Hz)
- "Known" co-channel interference

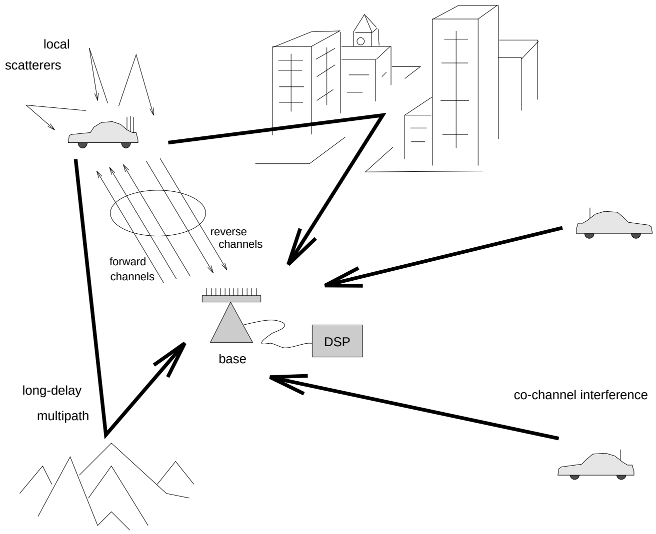


Figure 3.1: A typical mobile outdoor radio scenario

While researchers in communications have studied the use of adaptive arrays in indoor and outdoor environments over the past decade, the signal processing community has only recently become interested as military funding has dried up. Some of the current research thrusts in this area include the following:

- Blind adaptive arrays emphasized since the presence of numerous coherent multipath rays can render the use of array calibration data very difficult
- Multichannel equalization for mitigation of intersymbol interference, with and without training data
- Maximum likelihood approaches for taking advantage of a known (synchronous) signal environment to compute the array response and signal amplitudes
- Simple feedback schemes introduced to exploit the cooperative nature of such systems
- Capacity improvement studies conducted to determine the potential gain of using multiple sensors, primarily in outdoor mobile cellular radio
- Channel characterization studies performed for both indoor and outdoor environments, but typically only single channel measurements taken
- Optimum combining vs. maximal ratio combining performance comparisons

Technological and Research Challenges

Not surprisingly, the major hurdles in this area are those that are linked to practical issues such as how to use discrete array calibration data and how to implement complicated algorithms in real time. These problems have been well studied, but not solved. In addition, there are hordes of smaller practical problems that have not been addressed by the research community, and that are only now coming to light as systems such as those built by Arraycomm are tested in realistic signal environments.

A related issue is the characterization of the propagation channel in both indoor and outdoor environments. A large number of such studies have been conducted by the RF community, but what impact do their results have on the design of corresponding spatial and temporal signal processors for such environments? For example, will we expect to see diffuse multipath with broad angular spread, or is most of the energy concentrated in a few specular reflections? Is the multipath delay spread short enough so that coherence is maintained among signal rays, or will there be significant decorrelation?

It is clear from recent work by a number of researchers that a significant performance advantage can be obtained by exploiting the known structure present in communications signals, particularly with cooperative transmission systems. This began with the use of cyclostationarity, constant modulus, and decision direction for digital signals, but in many situations much more is known about the signal waveforms. Some examples of the challenges that could be addressed are

- Real-time algorithms, trade-offs between "optimal" approaches which are usually more computationally involved, and more efficient suboptimal methods
- Is it possible to develop methods that exploit the presence of timing, power control, and identification sequences within the information stream of an IS-95 signal?
- How can the periodic training sequences transmitted with GSM signals be properly exploited in multiple antenna systems for both spatial and temporal equalization?

- Since the base station and remote often communicate over several different channels simultaneously (eg, control channels for synchronization, pilot tones, and paging), is it possible to use (spatial?) information gleaned from a control channel with known transmissions in the traffic channel containing the unknown information stream?
- Is it feasible to use array calibration data? If so, how best to compensate for calibration errors?
- How much spatial DSP can be performed at the remote? How can one take advantage of the cooperative nature of wireless communications applications (eg, through the use of feedback between base and remote, etc.)?
- How can prior spatial information (eg, in the form of array calibration data) be combined with the type of prior temporal information listed above, so that all available structure in the problem is exploited?

Relevant Research at BYU

Our research group is studying the use of sensor array signal processing in wireless communications applications, both for outdoor mobile radio and indoor PCS systems. ArrayComm, Inc., has developed a working prototype of a mobile cellular radio transceiver that employs an eight element antenna array, and has agreed to participate in this research by providing data from a number of long range outdoor tests involving both AMPS and IS-54 signals. Our primary goal is the development and "live-data" performance analysis of multichannel algorithms for all aspects of the wireless communications problem, including symbol synchronization, channel equalization, diversity combining, and signal demodulation. One particular aspect of our work is the evaluation of whether or not performance can be significantly enhanced by exploiting array calibration data, even in situations where it is subject to errors. Our preliminary results and the success of the Arraycomm system indicate that imprecise calibration data can be reliably exploited and should not be ignored. While our NSF sponsored program is brand new, we have already obtained some interesting results in blind decision directed beamforming, and in the use of array calibration data in multichannel equalization. We are also exploring connections between the model used in calibrating a non-uniform linear array, and the model that results from attempting symbol synchronization and equalization in the frequency domain.

Decision Directed Beamforming

Idea is to exploit known (digital) modulation format of signals of interest (SOI). Assuming the availability of an initial (possibly crude) estimate of the SOI beamformer weights and symbol synchronization, initial symbol decisions are used to generate a reference (training) signal for computing minimum mean squared error (MMSE) weights $\mathbf{R}_{xx}^{-1}\mathbf{R}_{xs}$. This procedure may be repeated as often as necessary to improve performance, although it can be shown that asymptotically, only one iteration is necessary. We have conducted a comprehensive symbol error rate (SER) performance analysis of the algorithm. For example, SER expression for the case of a BPSK SOI is given by

$$P_{2d} = \frac{1}{2}\Phi\left(\sqrt{\frac{T \times \mathrm{SNR}_i}{\left[\left(\mathbf{A}^*\mathbf{A} + \sigma_n^2\mathbf{R}_{ss}^{-1}\right)^{-1}\right]_{dd}} - T}\right)$$

where $\mathbf{A}$ is the array response, T is the symbol period, σ_n^2 is the noise power, $\mathbf{R}_{ss}$ is the covariance of all signals present, SNR_i is the input SNR, $[\cdot]_{dd}$ is matrix element d, d, and

$$\Phi(x) = \frac{2}{\sqrt{\pi}}\int_x^\infty e^{-t^2}dt$$

is the complementary error function. Similar expressions have been derived for general M-ary PSK signals.

Figure 3.2 shows an example of the convergence rate of the decision directed method compared with other blind adaptive beamformers. The simulation involved a four element $\lambda/2$ uniform linear array (ULA) that received the waveforms of a 10dB QPSK signal at 10° and an 8dB Gaussian interferer at 14°. The decision directed method was initialized in this case by the CMA algorithm. The decision directed beamformer will also outperform beamformers that only use direction of arrival (DOA) information, as shown in Figure 3.3. In this example, the performance of the standard least-squares (pseudo-inverse) beamformer was compared with the decision directed method for the same scenario described above. The decision directed approach (initialized using the least-squares method with estimated DOAs) yields an order of magnitude lower SER even when the least-squares method uses the exact DOAs.

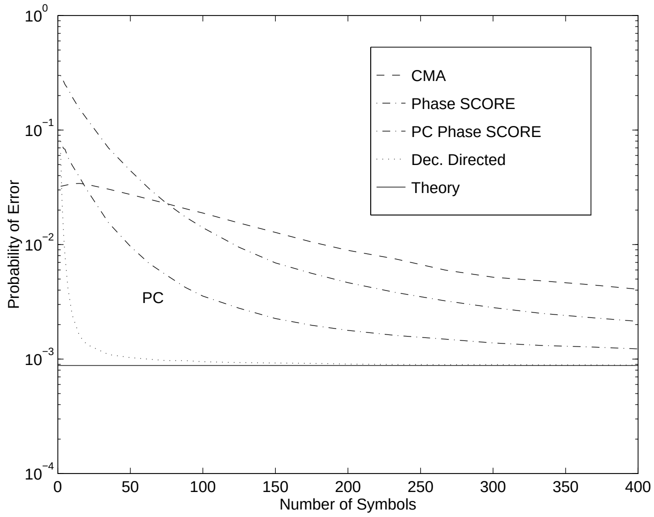


Figure 3.2: Probability of Error versus Number of Symbol Decisions Used

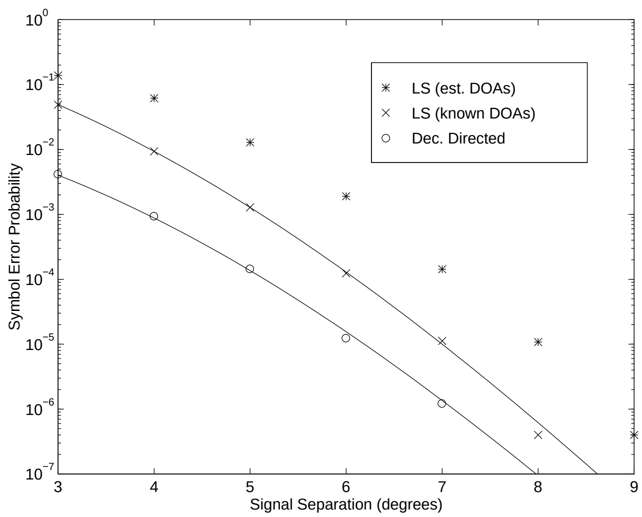


Figure 3.3: Probability of Error versus Signal Separation

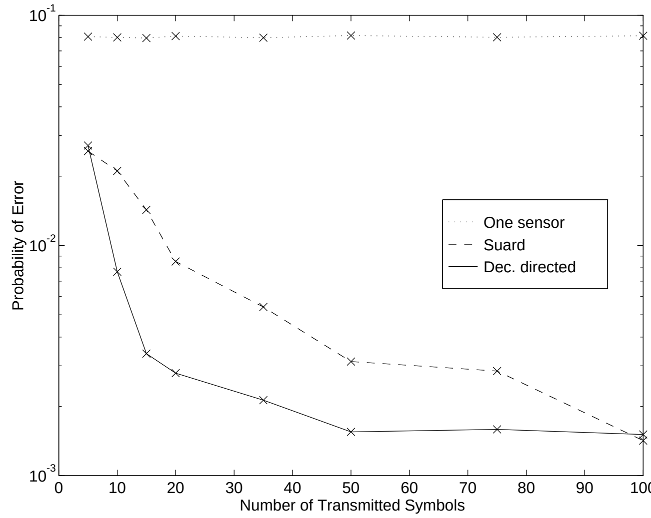


Figure 3.4: Probability of Error versus Number of Transmitted Symbols

The performance of the algorithm in CDMA signal environments has also been studied. Figure 3.4 shows the SER of the following three algorithms:

- Simple matched filter using only one sensor
- Recent algorithm of Suard, *et al,* (ICASSP '93)
- Decision directed, using single sensor decisions as initialization

in the following signal environment

- 5 element ULA ($\lambda/2$)
- long (2^{15}) QPSK spreading code with spreading factor of 64
- DOAs of signals uniformly distributed between $\pm 60°$
- random symbol/code offsets for each signal
- imperfect power control (log normal, mean 5 dB above noise, 2 dB standard deviation)
- SER results based on 10^6 bit decisions

Blind Multichannel Equalization

This work has focused on the tradeoffs involved in using array calibration data in multichannel equalization versus using unstructured FIR filter models between the source and antenna elements. The advantages of such an approach include a more concise parameterization of the problem, extra data structure imposed by the array response, and a potentially significant computational savings. On the other hand, there is no optimal "closed-form" solution, and the array calibration data is error prone. Our results indicate, however, that if it is available, the use of array calibration information can provide a significant performance advantage even if it is significantly perturbed.

As an example of this point, consider the results of Figure 3.5 which plots the performance of our equalization approach based on array calibration data, and that of a recent method by Liu, Xu, and Tong (which we have found to have the best performance among blind multichannel equalizers). This example involves a four element $\lambda/2$ ULA that receives a BPSK signal via a three-ray multipath channel. The direct path signal has a DOA of $0°$ and is at 20dB relative to the background noise, and one of the multipath rays is fixed at $10°$ and 12dB SNR, and is delayed one symbol period relative to the direct path signal. The other multipath signal has 19dB SNR, a relative delay of two symbol periods, and its DOA is varied between $0° - 30°$ over a number of different experiments. The root MSE signal estimate that results is plotted in Figure 3.5 for each of these experiments. The three separate curves for the "Array Method" correspond to three different levels of perturbations that were made to the nominal array calibration. The parameter σ_a represents the standard deviation of an additive complex Gaussian random perturbation that was independently made to each element of the array response matrix $\mathbf{A}$. Even in the case where $\sigma_a = 0.3$, which corresponds roughly to 30% gain and $18°$ phase errors, it is still clearly advantageous to exploit the calibration data instead of ignoring it.

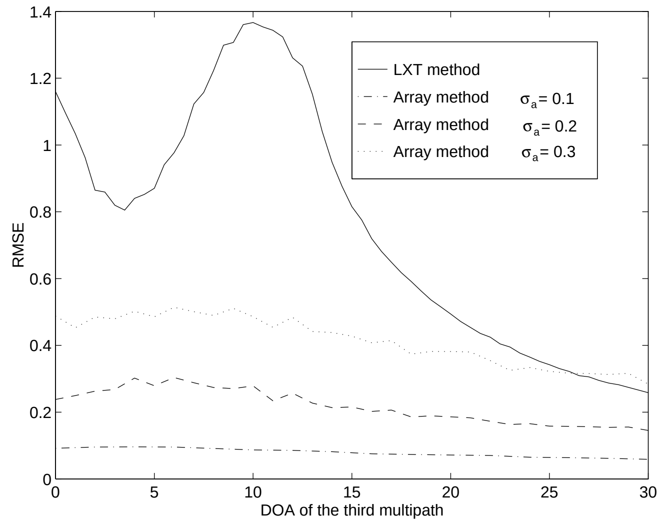


Figure 3.5: RMSE of Equalized Signal Estimate

Analysis and Mitigation of Array Calibration Errors

Additional related work at BYU has focused on how to characterize the effect of array calibration errors on direction finding (DF) and beamformer performance, and how to mitigate these effects. A summary of our results is given below:

- DF-Based Beamformer Performance Analysis
  - Comprehensive analysis of classical beamforming, least-squares (LS), total least-squares (TLS), principal components (PC), and structured stochastic estimate (SSE) beamformers.
  - Both MSE and SINR performance criteria used in analysis
  - Included effects of array calibration errors
  - $\text{MSE}_{SSE} \leq \text{MSE}_{LS} \leq \text{MSE}_{TLS}$, same for SINR
  - PC performance highly dependent on signal correlation
- Mitigation of Array Calibration Errors for DF
  - Gain/phase, mutual coupling, position errors considered
  - Conducted comprehensive analysis of existing DF algorithms
  - Developed on-line auto calibration techniques (MAP, LS)
  - Robust weightings for subspace fitting possible

Future Directions

Some areas of research that we plan to pursue are outlined below.

Frequency Domain Models

Consider the following model that represents the output of an arbitrary array in the frequency domain:

$$[\mathbf{x}(\omega_1)\ \ \mathbf{x}(\omega_2)\ \ \cdots\ \ \mathbf{x}(\omega_N)] =$$

$$\mathbf{A}\begin{bmatrix} e^{-j\omega_1\tau_1} & e^{-j\omega_2\tau_1} & \cdots & e^{-j\omega_N\tau_1} \\ e^{-j\omega_1\tau_2} & e^{-j\omega_2\tau_2} & \cdots & e^{-j\omega_N\tau_2} \\ \vdots & \vdots & \cdots & \vdots \\ e^{-j\omega_1\tau_d} & e^{-j\omega_2\tau_d} & \cdots & e^{-j\omega_N\tau_d} \end{bmatrix} \times \begin{bmatrix} s(\omega_1) & & & \\ & s(\omega_2) & & \\ & & \ddots & \\ & & & s(\omega_N) \end{bmatrix}$$

This model is appropriate for the following two situations:

- Problem #1: co-channel synchronization with d users

  Solution: All users transmit known waveform $[s(\omega_1)\ \cdots\ s(\omega_N)]$. If DFT frequencies are used, Vandermonde structure could be exploited by MODE, IQML, or ESPRIT to obtain computationally efficient estimates of delays $\tau_1, \cdots, \tau_d$. Estimate of array response $\mathbf{A}$ comes for "free." Once these parameters are determined, users may switch to unknown waveforms, and $\mathbf{A}$ estimate could be used to spatially isolate each signal.

- Problem #2: blind channel equalization for a single user

  All parameters identifiable (including array response) provided that $d < (N + m - 1)/2$. Array calibration can be used when few reflections are present, but no simple closed-form algorithm is available. Alternating projection would be a candidate approach here since signal and array response parameters are individually separable.

Beamforming using On-Line Recalibration

The idea in this approach is to use the given nominal calibrated response denoted by $\mathbf{A}(\theta, \rho_0)$, and track the actual response in the field $\mathbf{A}(\theta, \rho)$ by estimating "perturbation" parameters $\rho = \rho_0 + \tilde{\rho}$. The recalibrated response is then used in forming beamformer weights. There are a variety of available models for the perturbation **r$\tilde{\text{h}}$o** that include channel gain/phase imbalances, mutual coupling effects, position errors, etc. In addition, one could choose either of the following two approaches:

- Deterministic models (Friedlander, Paulraj, Ottersten, Kaveh, Ng, etc.) – requires identifiability of both **theta** and ρ
- Probabilistic models (Swindlehurst, Viberg) – identifiability condition relaxed assuming known prior distribution for $\tilde{\rho}$

In the latter case, an efficient *maximum a posteriori* estimate of $\tilde{\rho}$ has recently been developed for the asymptotic case (Viberg & Swindlehurst, IEEE Trans. SP, Nov. 1994).

Feedback for Spatial Diversity at Remote

The use of complicated DSP (eg, an adaptive array) at the remote is not considered feasible due to consumer cost. However, a feedback scheme could be developed in which the beamformer weights for the remote are computed at the basestation based on information fed back from the remote. These weights could then be transmitted back to the remote, where the only computational overhead required is the decoding of the weights from the information stream, and then applying them to do the linear combining. An outline of one possibility is given below:

1. Sequentially sampled remote array (only one digital channel needed)
2. Remote periodically resends received data from each antenna element with timestamp (retransmission rate ¡ few hundred Hz, depending on fading rate)
3. Base station computes remote beamformer weights using, eg, CMA or MMSE solution $\mathbf{R}_{xx}^{-1}\mathbf{R}_{xs}$.

4. Weights interleaved with forward channel traffic (or sent via a control channel) and picked off at remote

The above approach requires far less remote DSP than a recent scheme of Gerlach & Paulraj (Asilomar '93), and it is easier to interleave retransmission with reverse channel traffic in order to prevent downtime.

Interaction with Industry

An important component of our research will be technical interaction with ArrayComm, Inc., a small company studying the practical implementation of adaptive arrays in mobile outdoor radio. ArrayComm has developed a working prototype of a mobile cellular radio transceiver that employs an eight element antenna array, and has agreed to participate in our research by providing data from a number of long range outdoor tests involving both AMPS and IS-54 signals. Processing these live data sets will provide valuable exposure to many of the practical issues not normally addressed by the research community, and allow realistic testing of available adaptive array algorithms.

3.1.2 Blind Adaptive Beamforming for Digital Cellular/PCS Base Station Antenna Arrays

Michael D. Zoltowski, Purdue University

In mobile communications in an urban environment, the complex structure of the multipath propagation, and its rapid time-variation, make it impossible to have accurate information on the array manifold structure as a function of source location, as required by conventional adaptive beamforming techniques. The array manifold describes the relative amplitudes and phases across the array due to a unity power source at a given source location. If this information is not available, blind adaptive beamforming techniques are required. This has lead to the development of a number of blind interference cancellation techniques for both TMDA based and CDMA based cellular systems.

Blind Adaptive Beamforming for TDMA Digital Cellular

A new approach to blind adaptive beamforming for TDMA based cellular was taken which exploits the known symbol rate, the known pulse symbol waveform, and the known signal constellation, as opposed to exploiting spatial structure. The new approach is premised on multi-tone modulation, effected through subbanding via multirate DSP, so that the maximum multipath time delay spread is made negligible relative to the duration of the symbol pulse waveform in any sub-channel. This is done in a private cellular system called iDEN (integrated Dispatch Enhanced Network, formerly called MIRS) MIRS was developed and operated by Motorola and obviates the need for an equalizer. The new approach is simply based on the fact that the expected value of the magnitude square of each successive symbol period is the same irrespective of the timing offsets, the signal constellation, or the tails of the symbol pulse waveform. The PRO-ESPRIT algorithm previously developed by the PI with funding from NSF may be applied to a pair of spectral density matrices formed from the Fourier Transform of the expected value of the magnitude square of a symbol period evaluated at two distinct frequencies. The *i-th* generalized eigenvector of this matrix pencil is a weight vector which when applied to the array outputs yields the optimum signal to interference plus noise ratio (SINR) for the *i-th* source.

The array receiver structure and front end signal processing that produces the required input data for the new blind adaptive beamforming algorithm is shown in Figure 3.6. The full blown version of the algorithm, including extraction of timing information and exploitation of the signal constellation, is summarized via a flowchart in Figure 3.7 (a).

Initial simulations of the new approach to blind adaptive beamforming are extremely promising, revealing convergence rates much faster than current methods based on cyclostationarity. The best cyclostationarity based method to date for canceling co-channel interferers at the same data rate is the subspace-constrained Phase-SCORE method. As an example, Figure 3.7 (b) reveals that for a particular scenario the new method converges much faster to the maximum achievable Signal to Interference plus Noise Ratio (SINR) than subspace-constrained Phase-SCORE. The new algorithm converges in roughly 15-20 symbol periods for a moderate SNR scenario. Again, rapid convergence is important in highly variable multipath environments. The signal parameters are listed in the figure along with

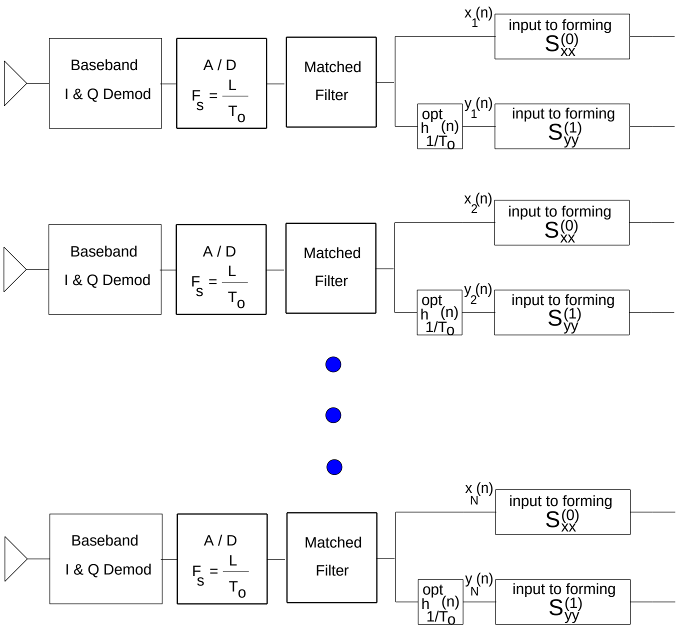
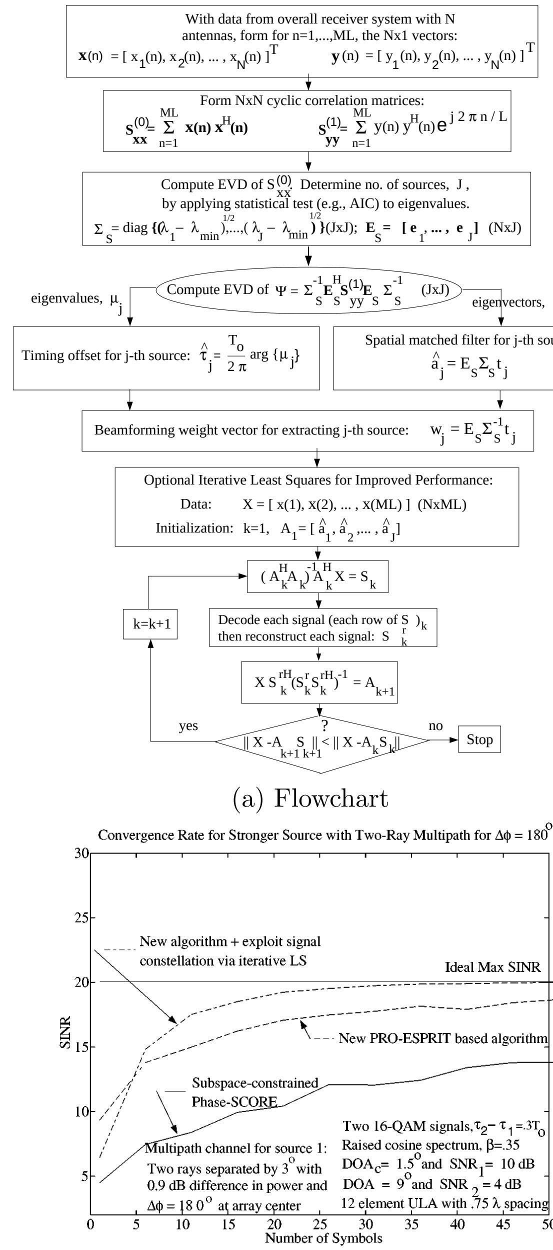


Figure 3.6: Front end receiver and signal processing operations for blind adaptive beamformer for TDMA with multi-tone modulation.



(a) Flowchart

(b) Simulation example

Figure 3.7: Flowchart and illustrative simulation example for blind adaptive beamformer algorithm for TDMA with multi-tone modulation .

the array parameters. An important characteristic to note is that the signal constellation was 16QAM so that the signals were not constant modulus thereby negating the use of algorithms premised on such. Multi-level signal constellations are needed to achieve high bandwidth efficiency in narrowband channels. Motorola's iDEN system employs a 16 QAM signal constellation.

In addition, although a simple two-ray multipath was used for source 1 in the simulation, it is important to note that the two paths were 180° out-of-phase at the array center. This is a worst case scenario from a SINR point of view. However, the angular separation between the two ray paths and the array aperture was large enough, respectively, such that there was enough diversity across the array for the algorithm to perform very well. In Motorola's iDEN system there is ten wavelengths of aperture across a horizontal structure that services a 120° sector of a cell. Thus, the parameters employed in this system are feasible relevant to current commercial cellular systems. Note that an interelement spacing slightly larger than a half-wavelength is allowed since the sector serviced is 120° in azimuthal width, as opposed to requiring end-fire to end-fire operation.

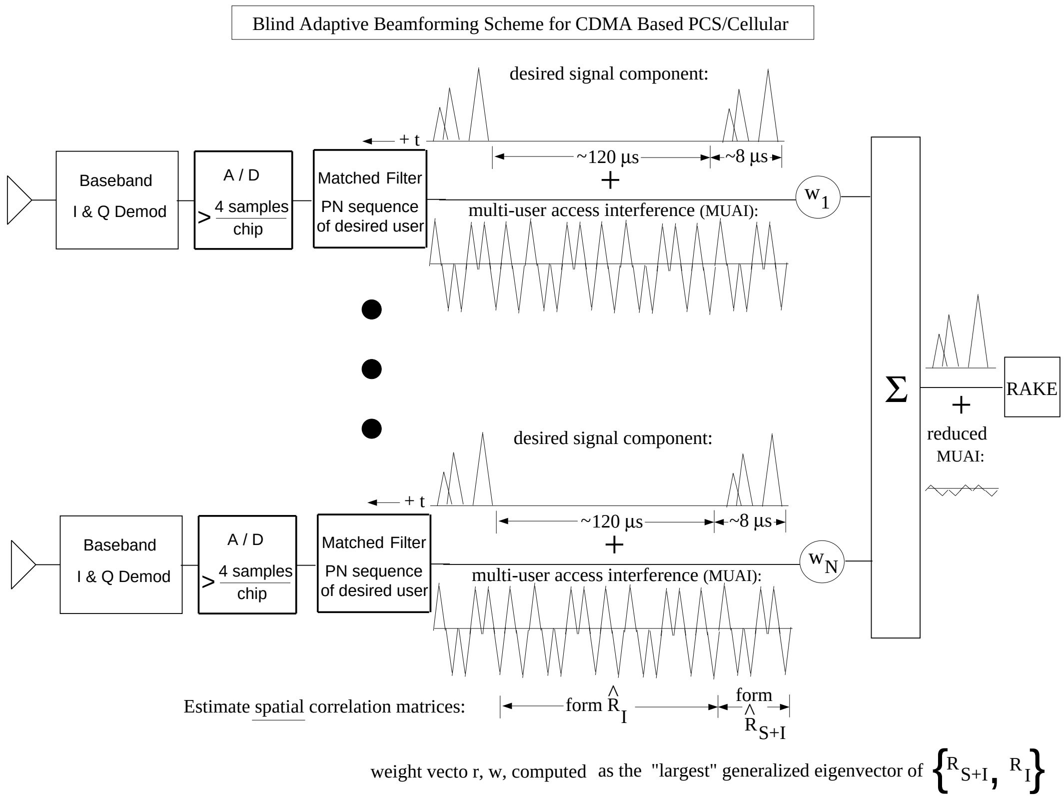


Figure 3.8: New blind adaptive beamforming scheme for CDMA.

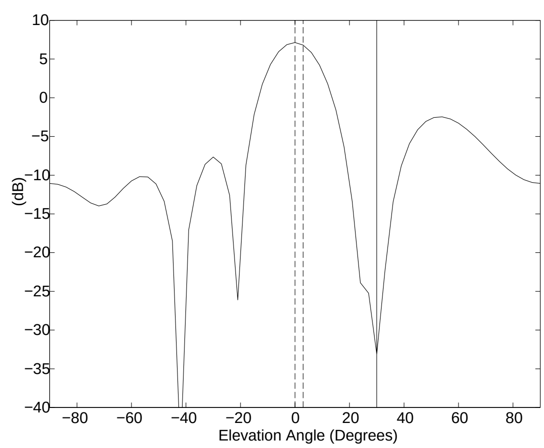


Figure 3.9: Illustrative simulation result for new blind adaptive beamforming scheme for CDMA.

Future Directions

Currently we are developing a blind adaptive beamforming scheme for a Direct Sequence CDMA Spread Spectrum cellular communications system that cancels co-channel interference, i.e., is resistant to the near-far problem, and optimally combines multipath. The near far problem occurs when the desired user transmits from the outer edges of the cell to the base, while another co-channel user with a different code simultaneously transmits very near to the base. The PN sequences assigned different co-channel users have some residual cross-correlation and for reasonably sized cells without power control, the output of the matched filter based on the code of the desired source is dominated by the user close to the base. Qualcomm has proposed power control schemes to combat this problem but these schemes are quite complicated and have been the primary obstacle in successful field tests of the IS-95 standard. Use of a smart antenna array allows one to implement a much looser power control for the same level of performance.

At the heart of the new blind beamforming strategy is a simple but powerful idea that is illustrated in Figure 3.8 for the case where the angular spread of the multipath is compact, i.e., less than half of a beamwidth. The idea is briefly explained below.

For each bit interval, the output of each antenna is put through a matched filter corresponding to the PN sequence of the desired user. In the IS-95 standard, this yields a 2 microsecond (in duration) triangular blip for each multipath arrival, for a given bit interval. Now, experimental measurements in an urban cellular environment reveal that the worst case time delay spread due to multipath is roughly 8 microseconds. At the same time, the bit period is roughly 128 microseconds in duration. Thus, in each 128 microsecond bit interval there is an 8 microsecond or so interval during which these 2 microsecond triangular blips occur corresponding to the different multipaths. This leaves at least 100 microseconds during which one can estimate the spatial correlation matrix of the co-channel interferers plus receiver noise, denoted $\mathbf{R}_I$.

The signal plus interference spatial correlation matrix, denoted $\mathbf{R}_{S+I}$, is estimated during the first 10 microseconds of a bit interval (and may be incoherently averaged over several bit periods.) Given $\mathbf{R}_I$ as well as $\mathbf{R}_{S+I}$, the criterion for determining the weight vector yielding the optimum signal to interference plus noise ratio (SINR) may be expressed as

$$\underset{\mathbf{w}}{\text{Minimize}} \ \frac{\mathbf{w}^H\{\mathbf{R}_{S+I}-\mathbf{R}_I\}\mathbf{w}}{\mathbf{w}^H\mathbf{R}_I\mathbf{w}} = \frac{\mathbf{w}^H\mathbf{R}_{S+I}\mathbf{w}}{\mathbf{w}^H\mathbf{R}_I\mathbf{w}} - 1$$

The solution is to choose $\mathbf{w}$ as the generalized eigenvector of the matrix pencil $\{\mathbf{R}_{S+I}, \mathbf{R}_I\}$ associated with the largest generalized eigenvalue.

The efficacy of the proposed blind CDMA beamforming scheme is illustrated via a simulation example. The simulation parameters were as follows. A vertical linear array composed of six elements equi-spaced by a half-wavelength was employed. Both the desired source and the interferer were DS-CDMA signals with different PN codes and 127 chips per bit; the modulation overlay was BPSK. Synchronization for the desired source was assumed. A simple two-ray multipath model was used for the desired source wherein the direct and specular paths arrived at elevation angles of 0° (broadside) and 3°, respectively, and with SNR's (per element prior to matched filtering) of 0 dB and -6 dB, respectively. The specular path signal arrived with a half-chip delay and 45° phase shift relative to the direct path signal at the array center. The interferer was simply modeled as arriving at a single discrete angle, 30° elevation, and was 20 dB stronger than the desired source prior to matched filtering, 4 dB stronger after matched filtering. $\mathbf{R}_{S+I}$ and $\mathbf{R}_I$ were estimated as described above with incoherent averaging over five successive bits during which the signal and interference characteristics did not vary.

The beam pattern obtained with the "largest" generalized eigenvector of $\{\mathbf{R}_{S+I}, \mathbf{R}_I\}$ is plotted in Figure 3.9. The beam pattern is observed to have near maximum gain in the respective directions of both the direct and specular path signals associated with the desired user, and have a deep null in the direction of the interferer.

We are also developing a scheme for estimating the relative time delay and angle of arrival of each triangular blip associated with the desired signal (each is associated with a different multipath transmission path) by applying the 2D ESPRIT algorithm, recently developed by the PI, to the difference matrix $\{\mathbf{R}_{S+I} - \mathbf{R}_I\}$. For small periods of time, this information is invariant from the uplink frequency band to the downlink frequency band and is thus useful for transmit beamforming on the downlink to avert the need for an adaptive antenna array at the mobile.

3.1.3 Summary of Smart Antenna Research at UT-Austin

Ganghuan Xu, University of Texas at Austin

Objectives

The objective of this research effort is to develop, implement, and validate advanced array signal processing techniques (or smart antenna technology) to significantly expand the channel capacity, improve the quality, and reduce the cost of various wireless communication systems.

Potential Impact

The demand of wireless communications is growing exponentially during the last five years and it is conservatively projected that by the year 2000 the number of users will rise up to 115 million nationwide. With such rapid growth, it is obvious that current cellular technology will be incapable of handling sufficient numbers of cellular phone calls simultaneously, because the spectrum allocated for mobile communications is limited. The smart antenna technology is the enabling technology to make significant expansion of channel capacity so as to accommodate the growing demand without requiring more bandwidth. Furthermore, a smart antenna system at a base station, can also significantly improve the quality of services, increase the coverage, and reduce the cost of the RF front end. Finally, due to powerful receiving capability of the smart antennas, the cost of handset can be significantly reduced and its battery life can be considerably increased. The ultimate impact to the society is that more and more people can enjoy the convenience of wireless communications at reduced cost.

Progress to Date and Milestones

Our progress in the smart antenna research can be classified into two areas: algorithm development and testbed development.

Algorithm Development

1. We have developed several blind equalization techniques that can successfully identify and equalize a multipath channel with a small number of symbols.

2. We have developed a new blind equalization based approach that can remove intersymbol interference and co-channel interference with one shot.

3. We have developed a new pre-equalization approach that can eliminate the intersymbol interference and co-channel interference in the downlink.

4. We have developed an integrated approach that can combine knowledge of the antenna array responses with the signal temporal properties to find the source directions-of-arrival more efficiently.

3.1.4 Theoretical Results on CDMA Detection Using Antenna Arrays

Mos Kaveh, University of Minnesota

Code Division Multiple Access (CDMA) is an attractive access method for modern wireless and personal communication applications. The performance of CDMA is limited by interference from other users. In particular, for a receiver which uses traditional matched filtering, strong (near) users can destroy the effective reception of desired weak (far) users, thus leading to the near-far problem. An attempt to mitigate this situation is to use power control. This, however, adds considerable complexity to the user transmitters and in some mobile cases may be impractical. A number of multiuser detectors have been developed to be completely or approximately near-far resistant. A question of interest is the degree to which the use of an antenna array at the base station can improve the reception of the desired user.

A strategy for answering the above question is to generalize the recently developed theoretical results on single channel multiuser detection to the multichannel (multi-element) case. This can be done based on a hierarchy of assumptions on the propagation statistics of the signals, and knowledge of the array response vectors. It is expected that such studies will illuminate the effects and parameters, through which, performance gains may be expected from the use of arrays. In the following, we summarize some results on such studies related to three multiuser detectors of varying complexity. The mechanisms which made an array improve the detector performances under fading and nonfading conditions are highlighted.

Array Decorrelator

The structure of an array decorrelating detector is show in Figure 3.10. $p_k(t)$ is the signature waveform for the k-th user, $\mathbf{a}_k$ is the array response vector for the k-th user and $\mathbf{R}$ is a spatio-temporal cross-correlation matrix

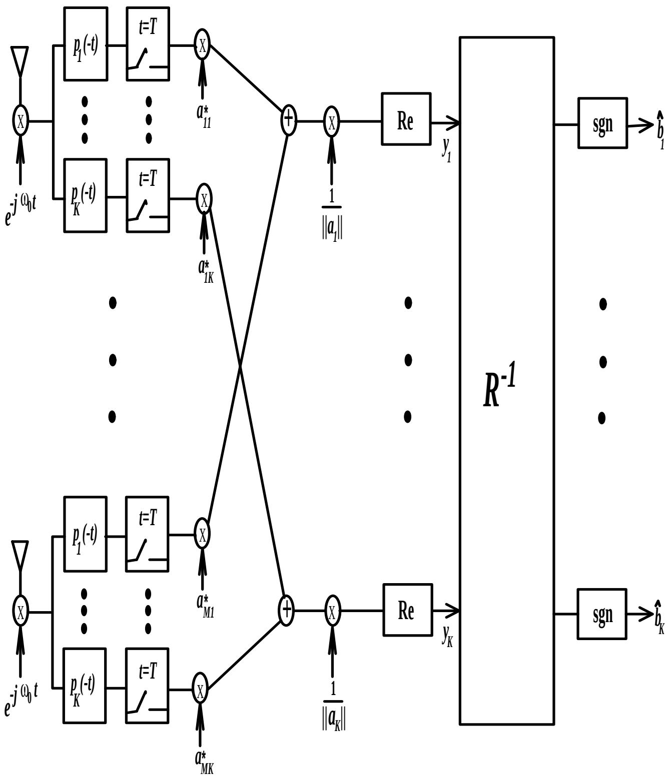


Figure 3.10: Multi-element decorrelating detector

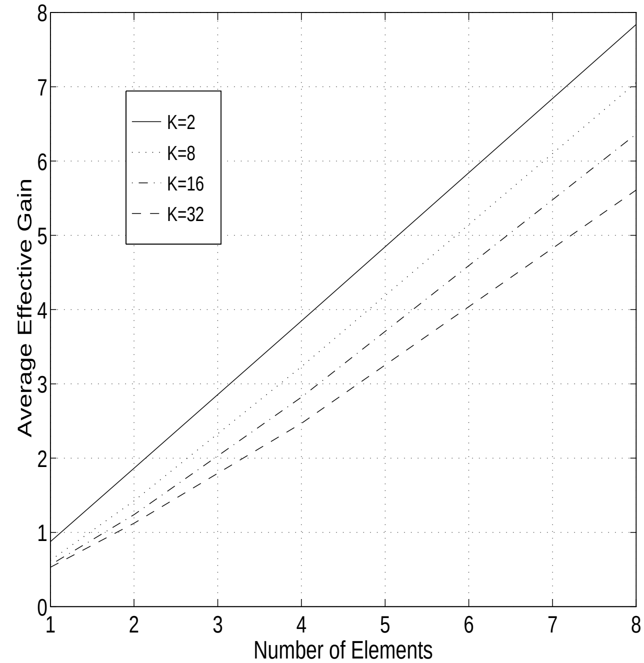


Figure 3.11: Average effective gain vs. array size

with

$$\mathbf{R}(k,l) = Re\left[\frac{\mathbf{a}_k^H \mathbf{a}_l}{||\mathbf{a}_k||\,||\mathbf{a}_l||}\right]\int_0^T p_k(t)p_l(t)\,dt.$$

If ρ_k is the k-th user signal energy-to-noise ratio, the probability of error for the k-th user, in a BPSK signalling scenario is given by

$$\mathcal{P}_k = Q\left(\sqrt{\frac{2\,||\mathbf{a}_k||^2\,\rho_k}{\mathbf{R}^{-1}(k,k)}}\right).$$

Clearly the performance of the array decorrelator is influenced greatly by the characteristics of $\mathbf{R}$, which in turn depends on temporal as well as *spatial* separations of the users. The use of an array reduces the Gaussian noise magnification accompanying the decorrelation process, in addition to providing classical array gain.

It is convenient to express the performance of a detector for a multiuser environment with that of an optimum detector operating in a single user/single antenna scenario. One such measure is the effective power gain, which measures the increase in the effective SNR over that for an optimum detector for a single antenna/single user environment. An effective gain of one, therefore, indicates that the array decorrelator has exactly compensated for the performance degradation caused by multiuser interference. Figure 3.11 is a plot of effective gain, averaged over uniformly angularly distributed interferences and uniformly distributed carrier phases, as function of the numbers of array elements (half-wavelength separated) and users, K.

It is clear that increasing the number of elements compensates for any correlation increases which may occur, by reducing the matched filter noise power.

Estimation of Array Response Vectors

When the array response vectors are known, an antenna array provides both white noise gain and increased discrimination between users. A more difficult problem is the case when the array response vectors are not known but must be estimated.This is the case in mobile radio environments where the motion of the transmitter causes the array response vectors to evolve in time. Several relevant questions are: (1) What types of multiuser detection strategies may be used when multipath fading is present? (2) What is the sensitivity of these structures' performance gains with respect to errors in array response vector estimation? (3) How can an antenna array increase the performance of multiuser detectors when the array response vectors are not known?

Two multichannel multiuser detectors have been studied for the fading environment. The first is an adaptation of the optimum multiuser detector which we shall term the decision-directed detector. It estimates the array response vectors or fading gains of all users based on past bit decisions and uses these estimates to optimally detect users. A second receiver is based on the decorrelating detector described above. It also bases its estimates on previous bit decisions and like the non-fading case has constant error probability regardless of the strength of interfering users. An analysis of the performance of these detectors shows that the error performance of the decision-directed detector is superior to the decorrelating detector. This superior performance however comes at the expense of increased complexity. The decorrelating detector requires only a linear transformation on the vector of matched filer outputs followed by bit slicing while the decision-directed detector requires the solution of an integer quadratic programming problem.

The second question may be answered by defining and evaluating a quantity termed *estimation efficiency* which measures the degradation in effective transmit power which takes place when only estimates of the fading gains are known. Evaluating this quantity for the two detectors above indicates that the decorrelating detector is less sensitive to estimation noise than the decision-directed detector. We conclude therefore that while the decorrelating detector has inferior performance overall, it is less sensitive to estimation noise.

When the array response vectors are known, it was shown above that the reduction in correlation between users signals which takes place, when an antenna array is used, lessens the noise enhancement inherent in the decorrelation operation. In addition the array provides white noise gain from adding the desired signal coherently and the noise noncoherently. In the fading case, when the array response vectors are not known, the array provides different advantages. The most important advantage appears to be spatial diversity. Spatial diversity lessens the effect of multipath fading by providing multiple channels which have fading gains which are at least partially uncorrelated. Weighting the antenna outputs by the their respective fading gains yields an output which has less fluctuation around the mean received signal power. Deep signal fades are therefore avoided. Besides spatial diversity gain, the multiple channels of antenna array provide an additional advantage when fading estimation is performed. Namely, estimation efficiency is improved as elements are added. In one numerical example based on a mobile array, the estimation efficiency was increased by 80 percent with respect to a single antenna.

Interference Cancellation with Multi-element Receivers

A relatively simple suboptimum detector, with approximate near-far immunity in the multiuser environment, can be formed as a generalization of the single channel interference cancellation receiver. Figure 3.12 shows the diagram for an array interference canceller. The basic idea is to detect the strongest user and cancel the interference caused by this user from the received signal. The strongest user is obtained by comparing the decision variables for all the K users. These correlation values are also used to find the order of cancellation for the different users. The detected bit is respread with the user's signature waveform and multiplied by its array response vector, and the result is subtracted from the received signal vector. The same procedure is repeated K times until all the users are detected.

It can be shown that while at each stage the multiple access interference from the strongest user is cancelled, a residual term is added to the total interference in that stage. This is due to nonzero correlations between the signature waveforms as well as array response vectors. It can be shown that this term is very small compared to the remaining multiple access interference.

Figure 3.13 shows the average probabilities of error as a function of the number of active users for a conventional matched filter detector and interference cancellation detectors using one element ($M = 1$) and two elements ($M = 2$). One can see the substantial reduction in probability of error from the addition of a second element, when the array response vector (two elements, in this case) is assumed to be known.

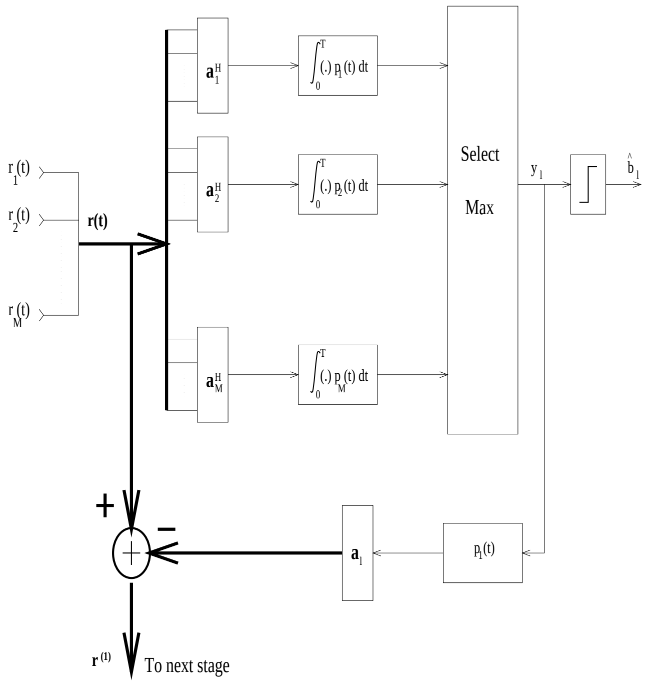


Figure 3.12: Block diagram of a multi-element interference canceller

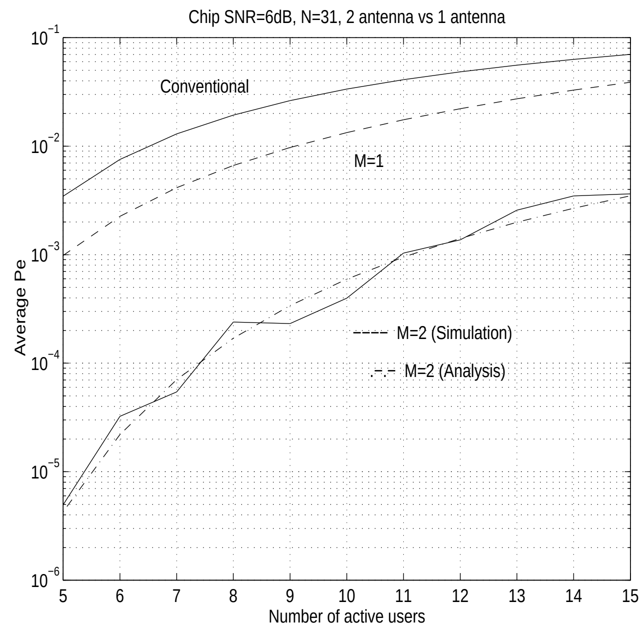


Figure 3.13: Average probability of error vs number of users-power control

Problem Areas and Future Directions

The idealized models, involving synchronization and knowledge of array response vectors have served the purpose of demonstrating, analytically, the best performance gains one may expect from the use of multi-element receivers in a multi-user CDMA environment. These results are encouraging. The results, however, point to the need for further study under more realistic operational conditions. In particular, future extensions will include:

- Investigation of the issues related to asynchronous systems.
- Design and evaluation of differentially coherent or other robust signalling and detection in the context of a receiver array.
- Inclusion of realistic models for array response variations and fading.

The above areas of work are very much predicated to results obtained from much needed studies which include:

- A need for significantly better spatio-temporal characterization of the signals in mobile urban and indoor environments with significant multipath content.
- Determination and classification of situations for which particular array processing strategies can be used to advantage. Space diversity combining is one simple and effective example which is in current use. Sectoring is another example.

3.2 Acoustic Array Processing

3.2.1 Processing of Microphone Arrays for Spatially-Selective Sound Capture

James Flanagan, Rutgers University

GOAL

To establish scientific understanding, engineering design, and prototype implementation for high-quality sound capture in reverberant, noisy enclosures.

POTENTIAL IMPACT

Using advances in digital signal processing and acoustic transducers, microphone arrays provide convenient, natural audio communication and sound reinforcement for large-group teleconferences, conference centers and meeting halls. For video/audio conferencing, they open opportunities to slave cameras to acoustic direction-finding algorithms.

PROGRESS

Recent research has established techniques for sound capture from selected spatial locations bounded in three dimensions. Matched-filter processing of arrays and multiple beam formation on a source and its images provide spatial volume selectivity. Array architectures include uniformly positioned sensors and randomly distributed sensors. Theory and simulation are confirmed by measurements in real rooms. A by-product is a software system for simulation of sound behavior in reverberant concave enclosures. Results for sound capture and sound location are being implemented in completely-digital real-time processor for a large-scale microphone array. Typical performance of a matched-filter array is shown in Figure 3.14.

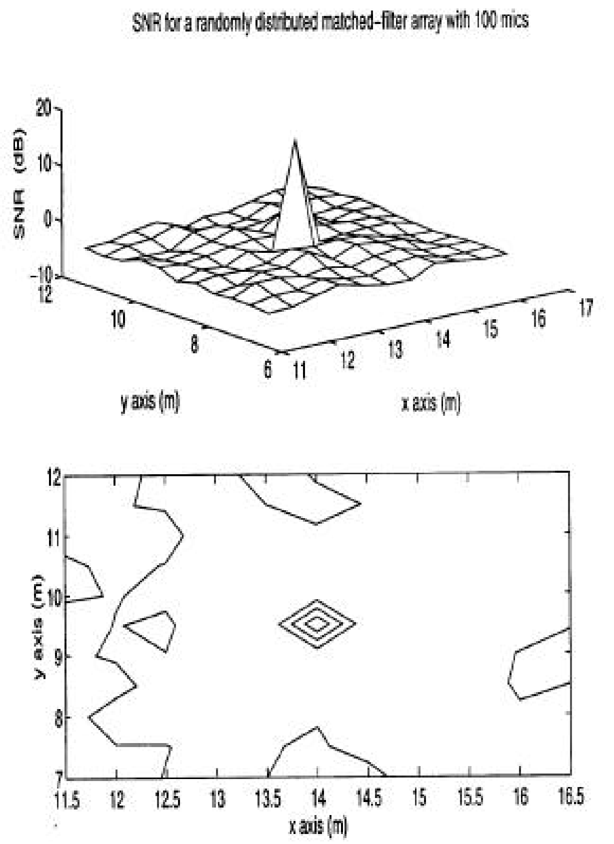


Figure 3.14: Signal-to-reverberant-noise ratio for sound capture by a randomly-distributed matched-filter array of microphones. The enclosure is a hard-walled room of dimensions (20×16×5) meters. The acoustic absorption coefficient is $\alpha = 0.1$. The array is composed of 100 microphones randomly distibuted on the (x,z) wall. The array is focused to coordinates (14.0, 9.5, 1.7) meters. The perspective plot illustrates the three-dimensional spatial selectivity of the matched-filter array.

TECHNICAL ISSUES

Continuing challenges include:

- accuracy of source location under severe reverberation
- optimal distribution of sensors as a function of enclosure geometry
- real-time processing for moving sources
- incorporation of diffraction effects into room simulations
- development of signal to noise measures that incorporate perceptual criteria

3.2.2 Calibration and Experiments For Signal-Subspace Detection Estimation Using an Ultrasound-in-Air Array

Mos Kaveh, University of Minnesota

This project has been a multifaceted approach to sensor array signal processing. Much of the research that has been carried out has been aided and/or motivated by experimental results obtained from a simple laboratory array testbed which uses ultrasound-in-air as the mode of signal transmission. Array calibration issues have been investigated and several methods of narrowband and wideband calibration have been developed. Experimental results have pointed to the practical limitations of signal-subspace detectors and estimators, leading to the theoretical analysis of the sensitivity of these detectors to model perturbation, and the development of an alternate detector and the design and performance analysis of a general class of MUSIC-like estimators with improved resolution threshold for calibrated arrays. Some of these results are summarized below.

Array Calibration and Experiments

The testbed consists of an array of eight piezoelectric elements, linearly and equally-spaced, operating at a carrier frequency of about 40 kHz at a bandwidth of about 2 kHz. The receiver has eight pairs of in-phase and quadrature (I/Q) channels, resulting in eight digitized complex baseband signals. For direction-finding experiments, the data is segmented and DFT'd, generating multiple snapshots of narrowband data at different frequencies. Up to three transmitting elements can be used, each of which is placed at a well-defined angle. Examination of the gain and phase characteristics of the receiver channels showed considerable deviation from the an idealized model for a linear uniform array. A summary of the calibration methods used are given below.

1. A single source is moved every 0.5 degrees. The estimated principal eigenvector of the sample covariance matrix, at a given frequency, is used as the array response vector, $\mathbf{a}_e(\theta_i)$. These vectors are stored in a calibration table. Such a table can clearly be used with methods such as a beamsum, maximum-likelihood and MUSIC for direction-finding.

2. Using K response vectors ($K \geq 1$), an $L \times K$ direction matrix $\mathbf{A}_e$ and an $L \times K$ ideal direction matrix for a linear uniform array, $\mathbf{A}$, are formed. The gain-phase calibration processor, $\mathbf{G}$, is determined such that:

$$\max_{\mathbf{G}} ||\mathbf{A} - \mathbf{G}\hat{\mathbf{A}}_e||_F,$$
$$\text{subject to: } \mathbf{G} \text{ being a diagonal matrix}$$

   Measured signal vectors are pre-processed by $\mathbf{G}$ before using algorithms, such as Root MUSIC or ESPRIT, for linear uniform arrays.

3. For calibration against gain, phase and mutual coupling, method 2 is repeated without the diagonal constraint on $\mathbf{G}$, and with $K \geq L$, to yield:

$$\mathbf{G} = \mathbf{A}\hat{\mathbf{A}}_e^H(\hat{\mathbf{A}}_e\hat{\mathbf{A}}_e^H)^{-1}$$

4. Calibration for angle-dependent errors can be done as an iteration on 2, or 3, based on fits in the neighborhoods of preliminary direction vector estimates.

5. Wideband calibration and focusing for a single-group of directions uses method 2 for each frequency to yield a focusing/calibration matrix $T_e(\omega_n)$.

$$\min_{T_e(\omega_n)} \quad ||\mathbf{A}(\omega_o, \beta) - T_e(\omega_n)\hat{\mathbf{A}}_e(\omega_o, \beta)||_F,$$
$$\text{subject to:} \quad T_e(\omega_n) \text{ being a diagonal matrix}$$
$$\beta: \quad \text{set of angles in vicinity of group angle}$$

Some of the results based on experiments are summarized below.

- Detectors such as MDL and AIC never worked with the experimental data. This is attributable to deviations of the data from the ideal models assumed for the formulations of these methods which are based on the eigenvalues of the estimated covariance matrix. The calibration methods which were attempted were not helpful in remedying this situation.

- Previously proposed autocalibration methods were not effective in dealing with the array characteristics.

- Calibration methods 1 and 2 have been reasonably effective with "spectral" and rooting based estimators, respectively. Method 1 has been generally more robust than 2.

- The estimation errors at high signal-to-noise ratios for all methods were consistently, considerably higher than predicted by the Cramer-Rao Bound. However, the relative performance of the estimators in terms of resolution, e.g. Min-Norm, Weighted-Norm MUSIC, Root MUSIC and ESPRIT having higher resolution than MUSIC, has been experimentally verified.

- Wideband focusing/calibration proved effective in dealing with coherent sources. These sources were generated by feeding two transmitters with the same signal.

- Active range-bearing estimation was demonstrated by using pulsed wideband signals, followed by range-binning and direction-finding per range bin. Calibration method 1, based on the passive calibration table was used in these experiments. Figures 3.15 and 3.16 show the received signal and range-angle map of three scatterers insonified by a pulsed transmitter placed above the array. MUSIC was used for angle estimation.

- Simple CDMA experiments, using two BPSK signals, were carried out which used adaptive beamforming to null the strong interference, without using a knowledge of the array response vectors. Figure 3.17 shows the output of a correlator for user 2, when a single element receiver is used and user 1 is 10 dB stronger than user 2. Figure 3.18 shows the correlator output when the array is used, and demonstrates the partial nulling of the interference, resulting in satisfactory detection of user 2.

Theoretical Results and Development of Detectors and Estimators

The experimental results pointed to the need for more robust detectors and higher- resolution MUSIC-like esti-

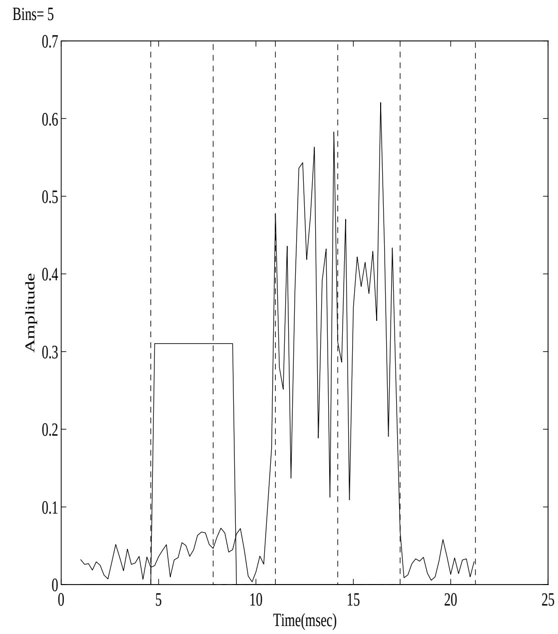


Figure 3.15: Demodulated output of one sensor, divided into 5 bins

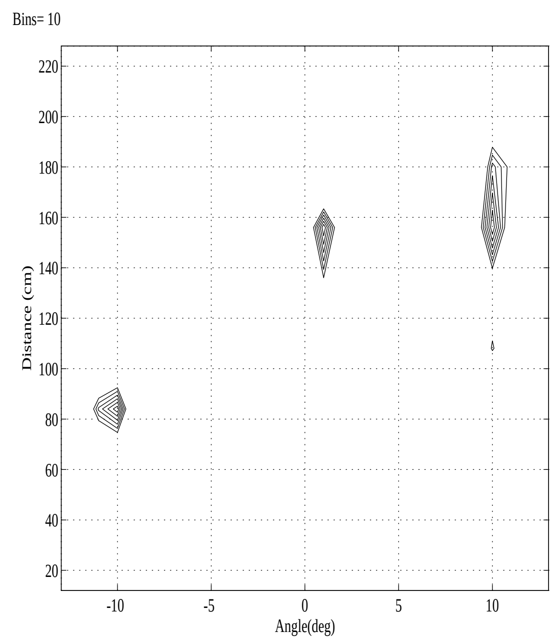


Figure 3.16: Range-angle map of 3 scatterers

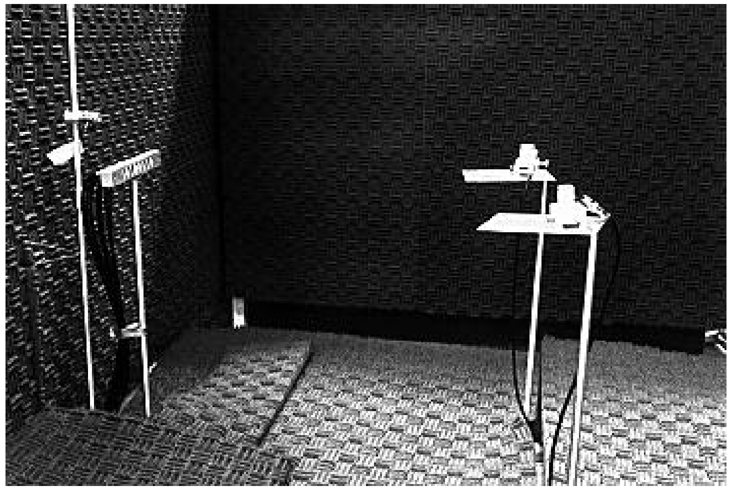


Figure 3.19: Experimental smart acoustic array system developed at the University of Minnesota.

mators which can operate with calibration tables of the type described in calibration method 1. A summary of the results follows.

- The negative experimental results of information-theoretic detectors motivated a comprehensive theoretical analysis of the performance and sensitivity of these detectors. Results of this study were reported in W. Xu and M. Kaveh in a June 1996 Signal Processing Transactions paper and verified the high level of detector sensitivity to model errors.

- Attempts at the design of more robust detectors, which can use array calibration data, yielded an eigenvector-based detector. This detector was reported on at the Sixth SSAP Workshop in Victoria, BC. The method has proved to be very robust in the experimental trials.

- The approach taken in the design of the estimators has been to provide an estimator structure with a desirable known asymptotic distribution, which can then be optimized based on the minimization of such large-sample properties as bias and resolution threshold. Two classes of MUSIC-like estimators based on the weighted-norm extension of MUSIC and a parametric generalized distance between particular vectors in the signal subspace, have been developed. The computational requirements of the latter are the same as for MUSIC. It has been shown that, both estimators can provide estimators with smaller bias than MUSIC at low SNR and with significantly improved resolution over MUSIC as demonstrated by theory, simulations and experiments, using calibration method 1.

Problem Areas and Needs

- A nagging issue in high resolution/sensitivity detectors and estimators continues to be the robustness of the algorithms against model errors. It is clear that not all algorithms can meet, or even come close to their theoretically predicted performance under all practical settings and propagation modalities. We believe that, from a practical point of view, the detection problem is still quite open. A good detector is essential, since detection is often the first step in the formulation of a signal-subspace estimator–an issue which is often ignored in the development and analysis of estimators.

- Basic algorithm development and performance analysis for direction finding has now reached some level of maturity. Performance measures, by necessity, have often been based on idealized models and asymptotic arguments, leading to claims of superiority of one method over another. This approach is

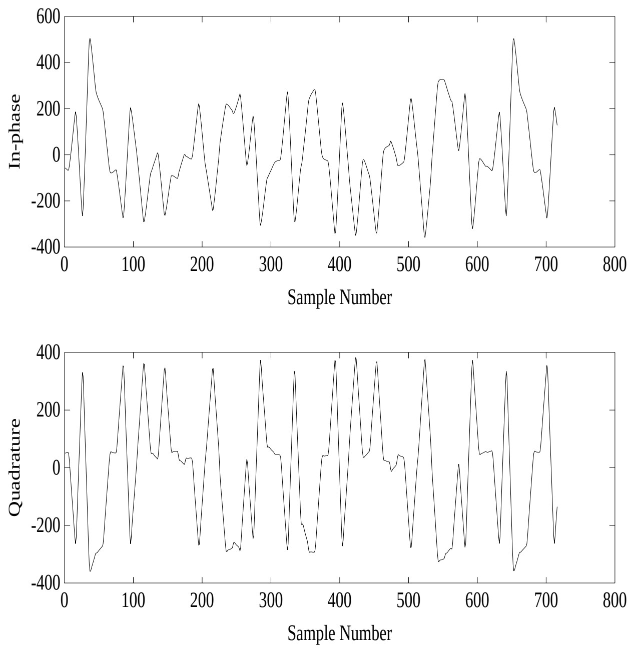


Figure 3.17: Correlator outputs for user 2, using one receiver element

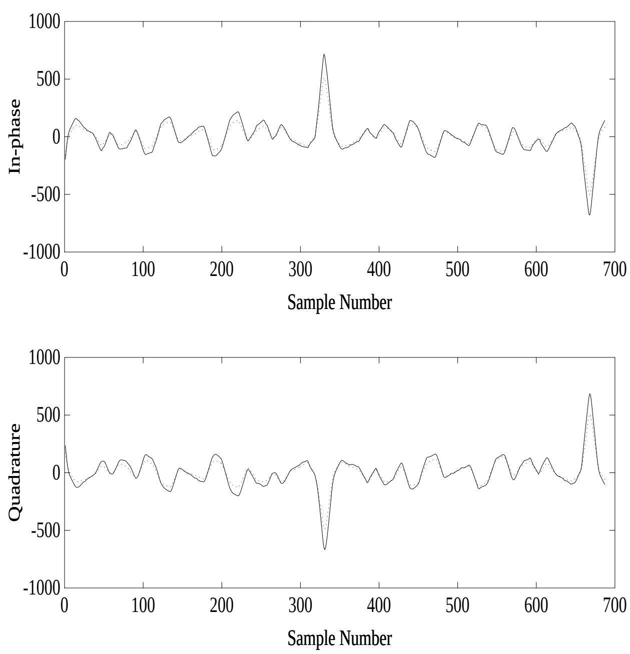


Figure 3.18: Correlator outputs for user 2, using the eight-element array

valuable and continues to be needed, but is clearly no longer enough. Practical performance limitations may be much more limited by such factors as calibration errors and the inherent robustness of a particular estimator to model assumptions.

- Hardware demonstrations and experimental systems are valuable for lending credibility to the methods which are developed. These systems take two forms: i) those designed for specific applications, ii) flexible testbeds. The first group of systems will continue to develop, usually in interdisciplinary efforts. The generic testbeds in the second group are needed to understand the characteristics of propagation and interference models, continued development of calibration methods and understanding of the limitations of algorithms under controlled and understandable conditions. Such testbeds may be developed collaboratively among several groups and serve as resource for algorithm developers.

3.2.3 Adaptive Microphone Arrays for Tracking and Beamforming

Eric Dowling, University of Texas at Dallas

The objective of this research is to develop hardware and software related to a system involving multiple microphones which are used to track a moving talker and to acquire his/her speech signal as he/she moves through a possibly noisy environment. This involves array processing methods such as direction finding to locate and track the speaker. It also involves beamforming techniques to acquire the signal. Also, the research addresses problems related to imperfections in any real world system as compared to the algorithmic modeling assumptions related to the system.

Experimental and theoretical research into adaptive microphone arrays is important because it will ultimately lead to vital commercial and military applications. Commercial applications include video teleconferencing, hands free cellular telephony, multisensor hearing aids, speech recognition front ends for advanced multimedia workstations, factory floor and noisy industrial communications applications, and passive motion detectors for burglar alarm systems. Military applications include hands free battlefield management oriented communications systems, passive acoustic alarm systems for front line operations, seismic direction finding arrays for vehicle detection and tracking, as well as other intelligence gathering and surveillance applications. The experimental research is important because it gives indications as to what kinds of errors in modeling assumptions most adversely affect performance. This sort of feedback guides research into developing more accurate and robust models that will support practical algorithms that will eventually be part of common technology. The problems encountered in experimental microphone array research are similar to those encountered in sonar, radar, and wireless communications arrays. Hence the research extends beyond microphone arrays into more general areas of array processing that deal with real world hardware and wave propagation with model uncertainties.

In order to study the problems associated with implementing such a system, we built an experimental test bed to collect data. The experimental array is pictured in Figure 3.20. The array electronics were designed for low noise and a high degree of matching between all channels. The front end of the system includes 32 electret microphones which are sampled (in a synchronized manner) at a rate of 16 KHz and 16 bits per sample. After initial amplification to 1 Vp-p at the microphone elements, the signals propagate through cables to the differential receiver board and then are filtered with 13 pole analog Butterworth anti-aliasing filters. These filters are closely matched and use 1 % tolerance parts. After filtering, the signals are multiplexed into a high speed A/D whose digital output stream can be routed to a hard disk for off-line processing or sent to a DSP processor through a TMS320 interface port for real-time processing. To date, we have primarily used the system in an off-line mode to collect data for algorithm analysis and verification.

As it turns out, the array points out some real world problems relating to array calibration and the effects of model imperfections on tracking and beamforming algorithms. Assumptions made by most array processing methods that are generally violated by a real system in a real world environment include: Channels have identical frequency response characteristics. The array is exactly steered (using delays across the channels so a signal from the desired location is time aligned across all channels) toward the speaker. The physical environment is non-reverberant.

Array calibration is fundamentally important to the proper operation of an experimental array. Once the system is built, the channels must be aligned. Also, even with careful design, the channels will not have identical amplitude and phase responses as a function of frequency. In a wideband application such as speech, this can cause difficulties for certain algorithms. From a software perspective, we would like our algorithms to be robust enough to handle some modeling imperfections. Even algorithms such as ESPRIT that do not require array calibration per say, do assume identical doublet subarrays. So channel mismatches can affect algorithms in various ways by violating modeling assumptions.

One objective of our research was to assess the ability of small to moderate sized arrays to use adaptive algorithms to significantly boost performance. As such, we implemented various forms of constrained adaptive beamformers such as the generalized sidelobe canceller (GSC), the soft constrained GSC, the Frost beamformer, and specially calibrated versions of GSC. Our findings indicate that channel mismatches and multipath signals due to the reverberant environment allow signal components to get past the constraint and hence much of the desired signal can be cancelled at the adaptive beamformer output. To alleviate these problems, we developed eigenvector based calibration schemes to enhance the performance of GSC operating under model uncertainties. The results were positive in this area.

Various beamformers were also used as a front end to a hidden Markov model based speech recognition system. These systems were tested with both simulated and real data from the experimental array. Results in certain noisy environments (e.g. fan noise interfering with a speaker) showed speech recognition rates moving from the 30% region to the 90% region.

In related work we applied several high resolution direction finding and tracking algorithms to moving sources in both non-reverberant and reverberant environments. These methods included MUSIC, ESPRIT, TLS, and Minimum Norm. Also the spherical subspace tracker was used to track nonstationary signals. Our findings indicate these algorithms worked fairly well in the nonreverberant environment, but not so well in the reverberant environment. Further, the algorithm performance seems to be limited more by modeling errors and multipath signal components than by anything else.

Several conclusions and ideas regarding future research directions emerged from this research. The per-

Figure 3.20: Experimental smart acoustic array system developed at the University of Texas at Dallas.

formance of many array processing algorithms is limited by channel mismatches and multipath propagation. Algorithms must be designed to take model mismatches and multipath propagation into effect. The eigenvector based methods we developed calibrated the array well and improved performance. More research is needed in the areas of on-line auto calibration and room acoustics / multipath channel equalization. Algorithms should be designed specifically to be robust to various model misspecifications as a design objective. Experimental work involving system development and algorithm performance must continue in order to facilitate technology transfer.

3.3 Array Design Issues

3.3.1 Array Design for High Resolution Imaging and Source Localization

Saleem Kassam, University of Pennsylvania

Introduction and Objectives

Imaging and source-locating systems are generally based on arrays of individual elements used to sense the propagating field produced by radiating or reflecting objects. Through signal processing techniques such as beamforming, an image of the object field may be formed. Passive arrays receive signals from radiation sources (emitters), whereas active-array imaging systems use arrays to both illuminate the object field and to record the reflected field. Examples of imaging array systems occur in radio astronomy, sonar, microwave imaging and radar, communications, and ultrasound imaging. Our research has been aimed at developing fundamental new results for array design and associated signal processing, based on recent developments on the characterization of array performance in linear imaging. Active imaging systems are a particular focus of our investigation, although our work also addresses passive arrays. One important area of application of new active imaging array concepts is in ultrasound imaging in medical and industrial settings. High-resolution emitter location is of much interest in acoustic and communication applications. New developments in active arrays are also of significance in such scenarios, where ideas of active arrays can be implemented in systems employing retro-transmission from emitters in the field in response to the received complex field from the transmit array.

An important concept in array design is that of the *coarray.* The coarray allows us to consider how to deploy array elements (and associated hardware) in the most efficient way to obtain large array apertures and high resolutions. This concept allows minimum redundancy active arrays and minimum complexity active arrays to be specified. Some of these ideas parallel those that have been developed in the past for passive arrays. The general concept of the coarray also provides a better understanding of imaging performance improvements that are possible with multi- frequency operation for both active and passive imaging. Additionally, new multiple-transmit or re-transmit and retro-transmit strategies can be investigated. The design of efficient arrays based on the coarray concept has an interesting connection with the problem of efficient design of finite-impulse-response digital filters with reduced computation requirements.

The definition of the coarray for narrowband, single-frequency operation is simple. For a passive array using correlation processing for incoherent arrivals, the coarray is the *set of location differences for all pairs of elements of the array.* This is called the *difference coarray,* and corresponds to the set of lags on which the autocorrelation function can be computed for the received field. It leads naturally to the idea of minimum redundancy passive arrays. The counterpart of the difference coarray is the newer idea of the *sum coarray* for active arrays. The sum coarray is defined as the *set of pairwise sums taken from the transmit and receive arrays* (which may be the same array). Consider the process of *generalized weighted linear beamforming* in which the transmit/receive elements are given complex weights to form a beam in a desired direction. The beam is scanned over all directions of interest to form an image. Three fundamental relationships exist for this type of imaging, which are

- The image is the convolution of the point spread function (PSF) with the reflectivity distribution of the scene,
- The PSF is the Fourier transform of a weighting function on the coarray,
- The weighting on the coarray is the sum of convolutions of sets of transmit and receive array weights.

Therefore, the sum coarray provides a geometric characterization of the class of PSFs that can be achieved by an active imaging system. A given array geometry can realize only those PSFs whose inverse Fourier transforms have support on the coarray. Similar results hold for passive imaging and the difference coarray.

This can be utilized to design minimum-redundancy active areas, and more generally minimum complexity active arrays under suitable definitions of array complexity. For multi-frequency or wideband operation, the coarrays are the *unions* of the coarrays defined at each frequency. The specific frequency used alters the coarray in that the locations are specified in terms of wavelength units. This provides a framework for analyzing the trade-off between number of frequencies or bandwidth and number of array elements needed to achieve a given level of performance in terms of resolution and sidelobes

The specific objectives of our recent work have been the following:

- Design Low Redundancy and Low Complexity Active Arrays in One and Two Dimensions
- Design Multi-Frequency Arrays and Analyze Array Performance for Wideband Operation
- Build an Audio Frequency Acoustic-in-Air Experimental Test- Bed to Validate New Array Concepts

Summary of Progress

We have obtained significant results for minimum redundancy line arrays for narrowband active imaging of reflecting objects or retro-transmitting sources. These lead to considerable reduction in front-end hardware requirements, without loss of image quality for quasistatic scenes. Alternately, these results can be used to improve the performance obtainable with given array hardware for a uniform line array without increasing hardware costs. The key idea exploited is that of the sum-coarray. The coarray of a filled uniform line array can be obtained with as few as one-third of the number of elements for typical array sizes of 64. Table 3.1 shows results obtained for minimum redundancy line arrays of lengths L up to 28. Here N is the number of minimum redundancy array elements needed for coarray equivalence to a filled array of L elements. The description of the arrays specifies the number of positions between physical element locations in the minimum redundancy array. Note that unit spacing here corresponds to half a wavelength. The measure R_a is a redundancy measure.

N	L	MRA	R_a
4	5	.1.2.1.	1.111
5	7	.1.2.2.1.	1.154
6	9	.1.2.2.2.1.	1.235
7	11	.1.1.3.3.1.1.	1.333
		.1.2.2.2.2.1.	
8	14	.1.1.3.3.3.1.1.	1.333
		.1.2.1.5.1.2.1.	
9	17	.1.1.3.3.3.3.1.1.	1.364
10	21	.1.2.1.5.2.5.1.2.1	1.341
11	23	.1.1.1.4.4.4.2.3.1.1.	1.467
		.1.1.1.4.4.4.4.1.1.1.	
		.1.1.3.2.4.4.2.3.1.1.	
		.1.1.3.3.3.3.3.3.1.1.	
		.1.2.1.2.5.2.5.1.2.1.	
		.1.2.1.5.2.2.5.1.2.1.	
12	28	.1.2.1.5.2.5.2.5.1.2.1.	1.418
		.1.2.2.1.7.1.7.1.2.2.1.	

Table 3.1: Minimum Redundancy Active Line Arrays

Longer arrays may be considered by restricting attention to *symmetric* arrays to reduce the computational burden. In this case we have obtained minimum redundancy arrays of length L up to 159. For such arrays, the redundancy is approximately 1.5, a major improvement over filled arrays that have redundancies of approximately L/4. Further reductions in hardware complexity for the front-end array elements can be realized by separately placing transmit and receive elements non-uniformly on a uniform grid, instead of collocating them. In fact, specific instances of hardware reductions of almost *an order of magnitude* can be realized over conventional uniform active arrays of realistic sizes. For example, a length 159 uniform line array can be replaced by a minimum redundancy symmetric active array of 31 collocated transmit/receive elements; a 163 element uniform active line array can also be replaced by an array of 19 transmit and 19 receive elements only.

For *multi-frequency* operation we have analyzed the passive imaging of incoherent sources and shown how performance improvements can be realized by designing arrays using the framework of the coarray for wideband operation. The simulation results of Figures 3.24-3.26

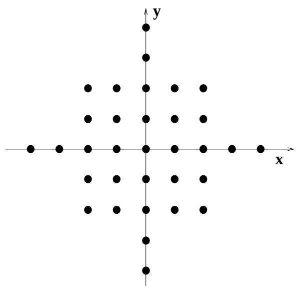


Figure 3.21: Coarray of 9-element Cross Array

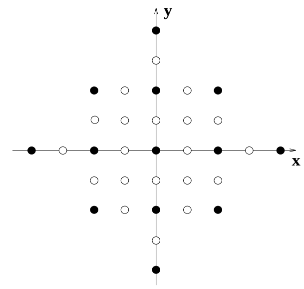


Figure 3.22: Coarray of 5-element Cross Array

for the coarrays of Figures 3.21-3.23 illustrate this. Figures 3.21 and 3.22 show the coarrays obtained for two cross arrays; the first is for a 9-element cross array with uniform ($\lambda/2$) spacing along the x- and y-axes, whereas the second is for a 5-element array with a spacing of λ. Thus Figure 3.22 is the difference coarray of the the 9-element cross array from which 4 elements have been deleted. Let us assume that the operating frequency here was 4 MHz. By operating the second array at three frequencies (2, 3, and 4 MHz) simultaneously, we obtained the more filled-in composite difference coarray of Figure 3.23. Corresponding point-spread functions for unity element weights are shown in Figures 3.24-3.26. The PSFs are plots of the region [-1,1]$\times$[-1,1] in **u**-space. The scale of the images is from 0 dB (white) to -35 dB (black). Figure 3.24 is for the original 9-element array at 4 MHz. Figure 3.25 corresponds to the coarray of Figure 3.22 at 4 MHz. The high sidelobes are a result of the holes in the array. Figure 3.26 shows the composite PSF, produced by operating the array with holes, at 2, 3, and 4 MHz. As expected, the composite PSF approaches that of the filled array as more holes are filled with multi-frequency operation. We are currently formalizing the design approach for arrays and element weights for multiple frequency operation to minimize array hardware and system complexity.

We have recently been able to obtain closed-form analytical expressions for the point-spread functions of active wideband arrays in reflection mode imaging, which allows performance and designs to be evaluated much more efficiently.

Our experimental acoustic array is being built out of low-cost audio- frequency components. The current operating frequencies are in the range of 5 to 10 KHz, and we have obtained 10-channel operation so far for

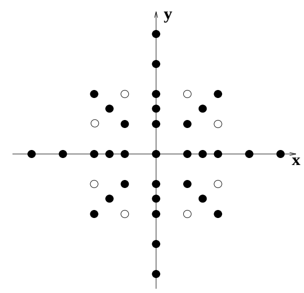


Figure 3.23: Composite Coarray for Frequencies 2, 3, and 4 MHz

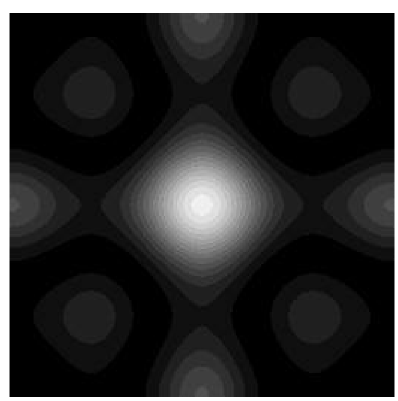


Figure 3.24: PSF for 9-element Cross Array at 4 MHz

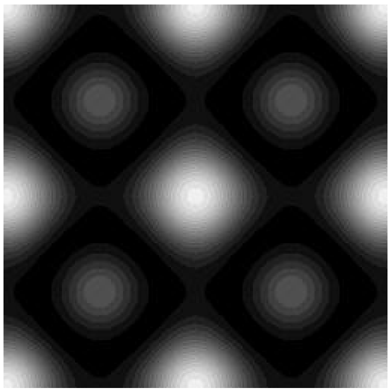


Figure 3.25: PSF for 5-element Cross Array at 4 MHz

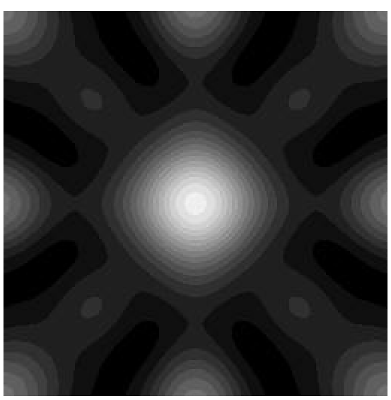


Figure 3.26: PSF for Composite Coarray at 2, 3, and 4 MHz

reflection mode imaging using 3-cycle pulses. Our plan is to develop this into a test-bed that will allow different array imaging and source location experiments to be performed with sparse (low redundancy) plane arrays, and allow investigation of the influence of array and processing imperfections on theoretical performance.

Conclusion

We are investigating the use of techniques of signal processing and array element configuration design that will allow minimzation of hardware cost and complexity of active as well as passive arrays for imaging and source location. Some recent work has suggested that it is possible to get an order of magnitude reduction in hardware with appropriate strategies for element placement in a fixed array for active imaging of a quasi-static scene. These results are related to work on minimum redundancy passive (correlation) line arrays, and for passive arrays also these ideas are of current interest for planar arrays. Associated with this idea of redundancy reduction is the use of multiple frequencies to further enhance the performance of imaging arrays, and to exchange elements for frequencies. This trade-off is made explicit by the concept of the wideband coarray of the array.

The current state of the art in active and passive array systems involves some *ad hoc* approaches, and the wideband mode is not very well understood from the point of view of imaging performance in conjunction with array design. Our work is currently aimed at addressing such issues.

3.3.2 Distributed Detection With Incomplete Knowledge

Rick Blum, Lehigh University

Objectives

Distributing sensors over a large area is necessary in some signal detection and tracking applications, particularly for surveillance systems. Such arrangements may even provide advantages over single sensor systems in terms of reliability, survivability, and improved signal detection performance. These performance improvements are the result of the inherent spatial diversity combining that occurs in such cases provided the sensors are separated by sufficient distances. In the interest of reduction of communication costs, simplification of processing, and preventing interception of one's communications, it is often advisable to use **distributed detection schemes**, which locate processors directly at each sensor. These processors reduce each sensor's observations to a multi-bit decision, and attempt to retain the essential information in these individual decisions needed to make a final signal detection decision.

Specifying the form of the partial processing at the sensors (the sensor decision rules) and specifying how the partially processed observations will be used in the final signal detection decision (the fusion rule) to obtain best performance is of fundamental importance. The need to specify each sensor's decision rule and the fusion rule makes distributed detection schemes inherently more complicated to design than the more common **centralized detection schemes**, where all observations are available in their original form at a central location.

Distributed signal detection has recently received significant attention, but the majority of this work has focused on signal detection problems where **a complete observation model is known**. The observation model characterizes how the signal that one is trying to detect affects the observations used to make the signal detection decisions and thus it is desirable for the observation model to be completely known. Unfortunately, in practice a complete description of the observation model is often unavailable. In our research, we are developing distributed detection schemes that will perform well when faced with such incomplete observation models. We consider cases with dependent observations from sensor to sensor, which arise frequently in practice. Due to their difficult nature, there has been very limited study of these cases by other investigators.

Summary of Progress

We classify our efforts into three basic areas: *unknown noise/clutter power results*, *unknown observation model results*, and *known observation model results*.

Unknown Noise/clutter Power

Part of our research has focused on the important and practical case of signals observed in additive noise-plus-clutter with unknown and possibly time-varying noise-plus-clutter power. This is a case of significant interest for many radar, sonar, communications, and medical signal processing applications.

Our research has focused on cases with dependent observations at the different sensors. Cases with dependent observations are known to be difficult and for this reason they have received very little attention. We have considered several different processing techniques and we have analyzed the advantages and disadvantages of each. Some specific design approaches have been outlined. In [4], we studied the cell-averaging (CA) constant false alarm rate scheme for distributed detection and in [1] we studied the order-statistic (OS) constant false alarm rate scheme for distributed detection. We found that the CA and OS schemes each have advantages. The OS scheme, for example, appears to provide better performance for cases with nonhomogeneous reference samples. In [4] and [1], only Gaussian noise-plus-clutter models were considered. In [2] we introduced a generalization of the CA and OS constant false alarm rate schemes which will provide better performance for cases where weak signals are observed in non-Gaussian noise-plus-clutter. The generalization requires a particular form of nonlinear processing of the observations. In [2] we demonstrate that the appropriate nonlinear processing can provide significant performance improvements in some cases with heavy-tailed noise-plus-clutter.

Unknown Observation Model

Cases with even less information about the statistical model for the observations are also of significant interest. In some practical applications it is possible that the probability density function (pdf) of the noise-plus-clutter may be unknown. For these cases, we have developed some nonparametric signal detection schemes based on the signs and ranks of the sensor observations [10]. These schemes provide constant false alarm probability for cases with additive random signal and noise-plus-clutter observations provided the noise-plus-clutter observations have an even-symmetric pdf. It has been demonstrated in [10] that the probability of detection achieved by these schemes is relatively insensitive to the exact pdf of the noise-plus-clutter observations. Since these schemes are based on signs and ranks they are usually less computationally demanding as well.

Another approach we are investigating is based on minimax robust statistics. We have already began applying these techniques to cases with possibly nonadditive noise models. Such cases are practical for many communication, radar and sonar system problems, particularly where nonlinearities act on additive signal and noise. There has been very little research on developing minimax robust schemes for cases with nonadditive observation models, so we initially investigated single sensor cases with multiplicative noise [7]. Some results were later obtained for more general nonadditive observation models [8]. Both [7] and [8] employ a performance criteria based on the asymptotic relative efficiency (ARE) of a detection scheme. So that minimax robust distributed detection schemes can be developed in a similar way, we recently investigated the use of ARE for distributed detection performance [5].

Known Observation Model

Sections 3.3.2 and 3.3.2 discussed our results for cases where some parameters describing the observation model were unknown. In each of these cases we were able to convert a problem involving optimizing the performance of a detection scheme when some parameters of the observation model are unknown to an equivalent problem of optimizing the performance of a detection scheme when the observation model is completely known. Thus our techniques require optimum schemes for cases with known observation models, so we have also made contributions in this area. This was necessary since there has been very little distributed detection research on optimum distributed detection schemes for cases with dependent observations, particularly for the Neyman-Pearson Criteria. We have obtained Neyman-Pearson optimum distributed detection schemes for weak signal cases and binary sensor decisions in [9], for narrowband signal cases with binary sensor decisions in [3], for multiple bit decision cases in [6], and for cases where signals may not be weak in [11].

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3.3.3 Maximizing the Accuracy and Resolution Capacity of Sensor Array Direction Finding

Yoram Bresler, University of Illinois

The research program of Yoram Bresler supported under the RIA grant focused on sensor array processing, and on image reconstruction. Research supported by the subsequent PYI award addresses four areas: (i) sensor array processing and parameter estimation; (ii) image reconstruction; (iii) optimal time-sequential acquisition of spatio-temporal signals; (iv) and vector field visualization. Significant results have been obtained in all these areas. In this report only the research on sensor array processing and parameter estimation is documented.

The project addressed the use of parametric models in high resolution imaging algorithms for sensor arrays, and had two primary objectives:

1. To assess and characterize the fundamental performance limitations in imaging processes using sensor arrays in terms of task-oriented quality measures.
2. To develop novel optimal algorithms for image acquisition and processing that exploit the underlying physical models for the signal formation and prior knowledge about the target, to optimize performance under adverse conditions.

The approach adopted in this project was to model the object in terms of a superposition of signals of known parametric form. Assuming a linear model for the measurement process, the problem then reduces to parameter estimation and detection for superimposed signals in noise. Specific goals that were addressed in this context were the following:

1. derivation of fundamental performance bounds on parameter estimation for superimposed signals.
2. Development of a design methodology for optimum array design.
3. Development of optimal and computationally efficient algorithms for parameter estimation of superimposed signals in noise.
4. Development of specific applications of these techniques to imaging problems.

Highlights of the results are given below.

Resolution Capacity of Wideband Sensor Arrays

A long standing limitation in sensor array processing has been the number of simultaneous directions (signals) that can be resolved by a given sensor array. This number, which we call the resolution capacity of the array, has been thought to be determined by the number of sensor elements in the antenna. In dense target environment, or in imaging applications where numerous superimposed signals are simultaneously present, this severely limits the performance of small arrays, such as those that can be mounted on an aircraft or a small vehicle. To address this problem, we studied the fundamental bounds on the number of co-channel wideband emitters resolvable by a passive sensor array. We showed that this number is only limited by the time-bandwidth-product of the observations, rather than by the number of sensors. In other words, if the signals are wideband (as they often are, in many applications) and the data is processed by an appropriate wide band processor (rather than by classical narrowband techniques), then by extending the observation interval, a theoretically arbitrary number of signals can be resolved. In practice, the number of actually resolvable signals will be also limited by their spacing and the noise level. This discovery, and the algorithm achieving these bounds, open a whole new realm of applications for small sensor arrays in communications, surveillance, and imaging.

Design of minimax-optimal sparse sensor arrays for multiple signal high resolution direction finding

As discussed earlier, in many applications, most notably in mobile systems, sensor arrays can only have a small number of sensors. In addition to the number of emitters that can be resolved by such an array, the achievable accuracy in determining the emitter directions is also affected by the .limited number of sensor elements. The first problem, of the resolution capacity of the array, was discussed earlier. The second problem, of maximizing the accuracy with a given number of sensors and given total size of the aperture (the available real estate) is the subject of this project. It involves a choice of a performance criterion, and an optimization technique to optimally place the available sensor in the aperture.

The design of such so called sparse arrays (using a reduced number of elements) has been researched extensively. The state of the art involves minimization of the level of secondary lobes in the beampattern (the directional response of the array to a single source.) Unfortunately, this classical criterion is not appropriate for optimizing the resolution of multiple targets using modern high resolution methods. As for the actual design procedure, a common technique has been random placement of the elements. Our work here is predicated on the premise that, by exploiting the nature of the problem and an accurate statement of the performance criterion, one should be able to do much better than random placement.

The performance criterion we use is the CRB on DOA estimates of up to m simultaneous sources *in the worst case scenario*. This is then an application of the results of the project on worse-case CRBs. We envision other such applications to the design of acquisition systems. By explicitly identifying this worst case, the design problem is reduced to a manageable search over the sensor positions. The resulting designs are far superior in resolving power to their classical counterparts.

3.4 Signal Subspace Algorithms for Detection & Parameter Estimation

3.4.1 Utilizing Sensor and Wave Properties for Antenna Array Processing

Jian Li, University of Florida

The parametric algorithms have high resolution but are sensitive to model errors. Yet in sensor array processing and its applications, model errors are often inevitable.

There are many challenges facing the current and future research efforts in the general area of array processing. It appears that most previous emphases have been on algorithms with the best performance and fastest computation speed. One of the major challenges for this area now is how to devise *robust* algorithms that have both excellent performance and fast computation speed. These algorithms should

- be based on appropriate data models,
- utilize sensor and wave properties,
- be insensitive to the estimated or assumed number of signals,
- be tested against both simulated and experimental data.

Another major challenge is to apply array processing algorithms to many practical applications including communications and radar. Due to the intense research efforts on array processing during the past two decades, many theories and algorithms have been developed and many insights have been gained. Armed with the knowledge we have learned, we can help make great progress in the aforementioned application areas. Yet the applications of array processing algorithms are not always straightforward. For example, in SAR (synthetic aperture radar) image formation and processing applications, it is a challenge to establish appropriate data models that incorporate electromagnetic phenomenology. It is also a challenge to coordinate multidisciplinary collaborations and to validate results obtained from experimental data.

Research Efforts at UFL

In collaborations with many researchers in the U.S. and abroad, especially with Professor Petre Stoica at Upsala University in Sweden, we have developed many algorithms for array processing and SAR image formation.

Array Processing

We have devised array processing algorithms

- for angle and waveform estimation in the presence of colored noise via RELAX (a RELAXation-based optimization approach),
- for efficient parameter estimation of partially polarized electromagnetic waves,
- for angle and polarization estimation with a COLD (Co-centered Orthogonal Loop and Dipole) array,
- for decoupled maximum likelihood angle estimation for signals with known waveforms.

We have found:

- It is better to obtain combined angle and waveform estimates. Concentrating out the signal waveforms, instead of making the estimation problem simpler, actually complicates the problem.

- Utilizing the properties of antenna sensors and the knowledge of incident signal waveforms can significantly improve the best performance that an antenna array can achieve.

For the topic of angle and waveform estimation in the presence of colored noise, we have described how the RELAX algorithm can be used for angle and waveform estimation of narrowband plane waves arriving at a uniform linear array in the presence of spatially colored noise. The RELAX algorithm is both conceptually and computationally simple; its implementation mainly requires a sequence of fast Fourier transforms. We have shown that the RELAX algorithm is an asymptotically statistically efficient estimator when the number of spatial measurements or the signal-to-noise ratio is large. The RELAX algorithm, however, is no longer an asymptotically statistically efficient estimator when the number of temporal snapshots is large. Both numerical and experimental examples have been used to demonstrate the performance of the RELAX algorithm for angle and waveform estimation. The experimental examples have also been used to compare the performance of RELAX with that of other well-known algorithms including ESPRIT with forward/backward spatial smoothing, MODE/WSF, and AP/ANPA. The RELAX algorithm is shown to be more robust than these existing algorithms due to the more relaxed assumption on the spatial noise and the simplicity of the algorithm. We have also shown that the RELAX algorithm is not sensitive to the assumed number of signals, which may be given or estimated.

We have applied the RELAX algorithm to the experimental data collected by the array system known as the Multi-parameter Adaptive Radar System (MARS). The array system was developed at the Communications Research Laboratory at McMaster University. The data was collected by deploying the array at the west coast of the Bruce Peninsula, Ontario, Canada, overlooking Lake Huron. MARS is a vertical uniform linear array consisting of $M = 32$ horizontally polarized horn antennas. The spacing between adjacent antenna sensors is 5.715 cm. The four sets of data we use below were collected when the array system was operated at frequencies 8.62, 9.76, 9.79, and 12.34 GHz. The data was recorded with 12-bit precision and sampled at 62.5 samples per second. For each carrier frequency, 127 snapshots were collected at each antenna output. There are two incident signals, a fact assumed to be known to the angle estimation algorithms. One of the incident signals is the direct path and the other is the specular path, which is reflected from the lake. The direct incident signal is a continuous wave (CW), whose amplitude is a constant and whose phase is a linear function of time. Since the specular path is a delayed and reflected version of the direct path, the phase difference between the two paths is a constant, which is determined by the time delay between the direct and specular paths, the carrier frequency of the waves, and the reflection coefficient of the lake. The incident signals arrive from near the array normal, but the exact incident angles are unknown since the vertical array structure may have been on a slight tilt. The parameter estimation algorithms we consider below do not assume any *a priori* knowledge of the incident angles and the signal waveforms.

Figure 3.27 shows the angle estimates obtained by using a single snapshot at a time as a function of the snapshot number for the direct path. (The results for the specular path are similar.) (Note that for Figures 3.27(a) – (c), not all angle estimates show up in the figures because they are either too large or too small.) The means and standard deviations of the angle estimates in Figure 3.27 were calculated by averaging the angle estimates obtained from all 4 carrier frequencies and all 127 snapshots. The RELAX algorithm is compared with the ESPRIT algorithm with forward/backward spatial smoothing using 10 sensors per subarray. RELAX is also compared with the MODE/WSF algorithm and the ANPA algorithm, which is related to the AP (alternating projection) algorithm. Note that the RELAX algorithm gives the smallest standard deviation for the angle estimates. Note also that although the ANPA algorithm minimizes the same cost function and is computationally more expensive than the RELAX algorithm, ANPA performs worse than RELAX.

Figure 3.28 shows the waveform estimates obtained by using RELAX with a single snapshot at a time when the carrier frequency is 8.62 Hz. (The results for other carrier frequencies are similar.) (All other algorithms yield very poor waveform estimates since they all estimate the incident angles first and hence they encounter a matrix ill-conditioning problem due to using occasionally very closely spaced angle estimates for waveform estimation.) Note that as expected, the phase difference between the estimated waveforms is nearly a constant for all carrier frequencies and the amplitude estimate is also nearly a constant.

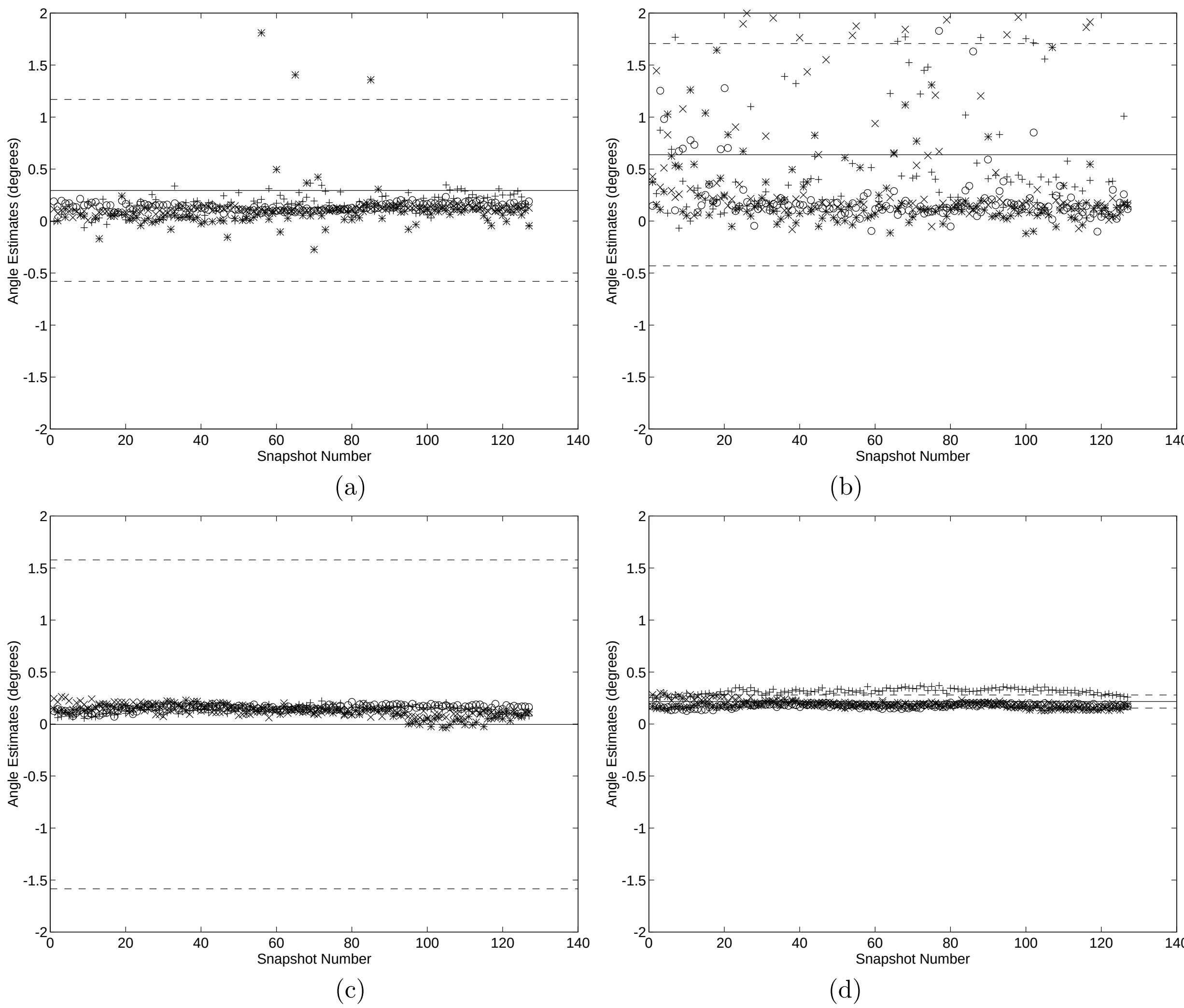
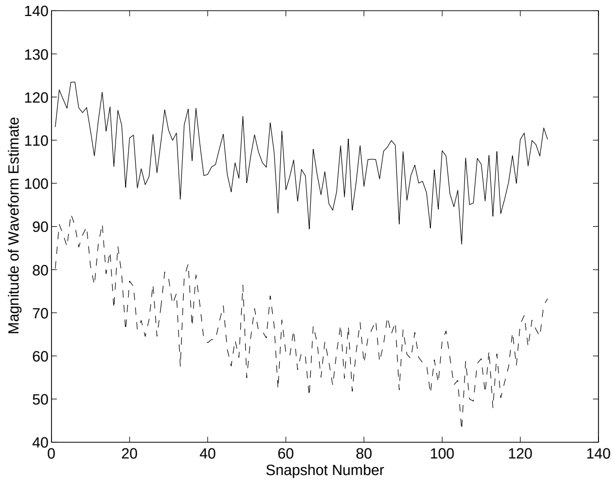
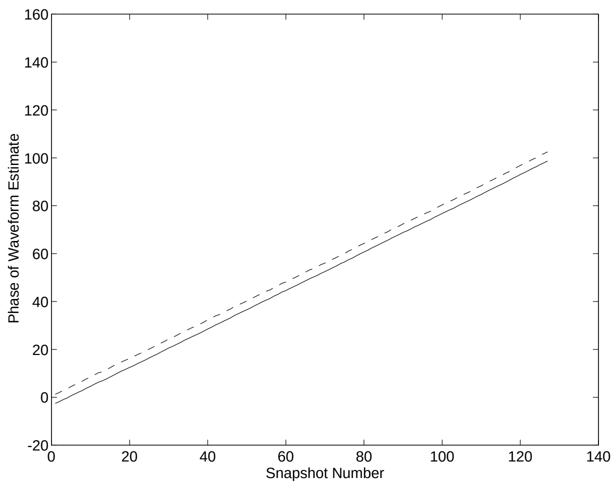


Figure 3.27: Angle estimates obtained from the experimental data collected with MARS as a function of snapshot number. The symbols "+", "×", "∗", and "o" are for the carrier frequencies 8.62, 9.76, 9.79, and 12.34 GHz, respectively. The solid lines denote the means and the dashed lines denote the means plus and minus the standard deviations of the angle estimates. (a) – (d) are for the direct path and are obtained with ESPRIT, MODE, ANPA, and RELAX, respectively.



(a)



(b)

Figure 3.28: Waveform estimates obtained by applying RELAX to each snapshot of the experimental data collected with MARS. The solid and dashed lines are for the direct and specular paths, respectively. (a) and (b) are for the amplitude and phase of the waveform estimate, respectively, when the carrier frequency is 8.62 GHz.

For the topic of efficient parameter estimation of partially polarized electromagnetic waves, we have considered the problem of statistically efficient estimation of the parameters of partially polarized electromagnetic (EM) waves with a uniform linear array of crossed dipoles. Previous research considered only completely polarized EM waves. We consider the maximum likelihood (ML) estimation of partially polarized wave parameters, in particular, the incident angles and the degrees of polarization. The polarization state of a partially polarized EM wave is a function of time while a completely polarized wave has a fixed state of polarization. Partially polarized waves can be found in many applications such as radar and ionospheric radio. For example, during the observation time, the state of polarization of a radar return received by a radar with polarization diversity can vary even though the original transmitted wave is completely polarized. This variation occurs because of the nonstationary behavior of targets, clutter, and other disturbance sources. We have devised a computationally efficient large sample ML estimator that avoids the multidimensional search over the parameter space, which is required by the exact ML estimator. We have also considered how to deal with the cases where some of the incident waves are known or are considered to be completely polarized.

For the topic of angle and polarization estimation with a COLD array, we have shown that by using the COLD array, the performance of both angle and polarization estimation can be greatly improved as compared to using a crossed dipole array. We have presented an asymptotically statistically efficient MODE algorithm that can be used with the COLD array for both angle and polarization estimation of correlated (including coherent) or uncorrelated incident signals. We have shown with numerical examples that the estimation performance of the MODE algorithm is better than that of the MUSIC and the noise subspace fitting algorithms, especially for highly correlated incident signals.

For the topic of angle estimation for signals with known waveforms, we have devised a large sample decoupled maximum likelihood (DEML) angle estimator for uncorrelated narrowband plane waves with known waveforms and unknown amplitudes arriving at a sensor array in the presence of unknown and arbitrary spatially colored noise. The DEML estimator decouples the multidimensional problem of the exact ML estimator to a set of one-dimensional problems and hence is computationally efficient. We have derived the asymptotic statistical performance of the DEML estimator and compared the performance with its Cramér-Rao bound, i.e., the best possible performance. We have shown that the DEML estimator is asymptotically statistically efficient for uncorrelated signals with known waveforms. We have also shown that for moderately correlated signals with known waveforms, the DEML estimator is no longer a large sample maximum likelihood (ML) estimator, but the DEML estimator may still be used for angle estimation and the performance degradation is small. To estimate the arrival angles of desired signals with known waveforms in the presence of interfering or jamming signals, we have shown that modeling the interfering or jamming signals as random processes with an unknown spatial covariance matrix may give lower CRB than modeling them as unknown deterministic incident signals.

3.4.2 Subspace Applications in Array Processing

Barry Van Veen, University of Wisconsin

Research Program Summary

This research program has focused on mapping data into subspaces prior to application of signal processing algorithms for the past six years. The subspaces of interest are generally not data adaptive, but are fixed a priori. The signal processing algorithms studied include adaptive beamforming and filtering, adaptive detection, and spectral analysis for time series and sensor array data. Recently the research has emphasized nonlinear filtering and non-Gaussian signal processing problems. Subspace signal processing algorithms generally exhibit improved performance for short data record situations by reducing the number of statistics that are estimated from a given number of data records. Computational complexity is often significantly reduced by subspace processing. Furthermore, subspace algorithms are generally less sensitive to model mismatch. There is usually a large data record or asymptotic performance loss associated with the discarding of information; however, this loss can be minimized by appropriate subspace design procedures.

The general approach is to first establish a subspace mapping framework for the signal processing algorithm of interest, and then to assess the algorithm's performance as a function of the subspace. This performance assessment is used to establish criteria for subspace design in the problem of interest. Lastly, a subspace that optimizes the design criteria is determined. The following paragraphs describe progress in specific areas.

In adaptive beamforming the goal is to reduce the number of adaptive degrees of freedom. This reduces the computational burden by decreasing the size of the covariance matrix that must be estimated and inverted to determine the adaptive weights. o We have shown that the adaptive convergence rate improves as a consequence of the reduction in adaptive degrees of freedom. o New techniques for designing the subspace transformation to minimize the steady state interference cancellation loss have been developed. o A dimension recursive modular structure for implementing reduced dimension beamformers has been derived. Analogous to a lattice filter, this modification of the generalized sidelobe canceler efficiently evaluates the output of beamformers having differing number of adaptive degrees of freedom by computing the output of a beamformer with p adaptive weights from the output of a beamformer with p-1 adaptive weights. o Lastly, we have shown how to design the dimension reducing transformation to significantly reduce or eliminate the signal cancellation caused by correlation between broadband signals and interference. Here we discard adaptive degrees of freedom used to cancel the signal and retain those used to cancel interference.

Subspace based minimum variance spectrum estimation has also been investigated. This problem is very similar to adaptive beamforming and thus many of the adaptive beamforming contributions have been extended to this problem and are not described further. One significant effect of subspace processing is to decrease the variability in the spectrum estimates computed from short data records.

Subspace methods have been applied to adaptive detection of deterministic signal having unknown parameters, such as complex amplitude, in Gaussian noise and interference with unknown covariance. The detector studied is adaptive in that it estimates the unknown parameters from the data and substitutes them into the likelihood ratio according to the generalized likelihood ratio principle. Here we studied mapping the data into a reduced dimensional space prior to performing detection. This detection problem has been addressed for both conventional sensor arrays and arrays composed of electromagnetic vector sensors. Milestones in this area are several: o We have shown that the same basic detector form applies to conventional and vector sensor data and signal models for narrowband and broadband vector sensor data have been developed. o A new form of the detection statistic has been derived that is computationally efficient for the subspace detector. o The subspace detector performance is shown to greatly exceed that of the full space detector when small numbers of data records are available. This is a consequence of the improved statistical stability of the estimates of the reduced number of noise covariance matrix elements estimated in the subspace detector. o Methods to design subspace transformations that approximately minimize the asymptotic (large numbers of data records) detection performance loss have been developed. o We have shown that the subspace detector is less sensitive to mismatch between the actual and assumed signal models. o A correspondence between subspace adaptive detection and reduced dimension adaptive beamforming has been established. o A modular structure has been developed that efficiently evaluates the detection statistic for subspaces of different dimension. o The subspace detection framework has been used to compare the performance of conventional sensor and vector sensor detectors with equal numbers of elements. The vector sensor array is advantageous with relatively large numbers of data records or when the scalar sensor is partially or completely blind to the signal to be detected. However, a subspace vector sensor detector offers both the signal sensitivity of a vector sensor array and the smaller dimension of a scalar sensor array, resulting in both the greatest flexibility and potential performance. o Several methods for designing subspace transformations specifically for vector sensor detectors have been developed.

Our most recent work has emphasized subspace applications in non-linear filtering and non-Gaussian problems. The nonlinear filtering work has applied primarily to time series to date and is not discussed further here. The non-Gaussian aspect has focused on subspace representations for higher order statistics such as the third and fourth order cumulant matrices. Again, time series applications of this work are not reported here. o As in other problems, the subspace formulation reduces the computational burden by reducing problem dimension. This reduction can be particularly dramatic due to the inherently large dimension of these problems. o Several methods for designing subspaces that minimize the representation error have been derived. o We have shown the error in a sample third order cumulant matrix estimate due to Gaussian noise is reduced by approximately the cube of the ratio of subspace to original dimensions under low SNR conditions. o Bounds on the error incurred with a subspace representation of data have been derived.

Description of Major Research and Technological Challenges

The primary challenges in the application of subspace techniques appears to lie in non-Gaussian and nonlinear signal processing problems for detection and estimation. The potential benefit of subspace methods in these areas appear to be much more significant than in problems based on second order statistics of the data.

Perhaps the greatest challenge for subspace techniques and array processing in general lies in adapting general principles and methods to solve specific application problems. This implies a shift from development of generic array processing tools or algorithms to applications of the tools developed over the last twenty five years. Two examples of specific applications include communication problems where a particular modulation scheme and channel characteristics are known and the biomedical problem of localizing sources of electrical activity in the brain from scalp measurements of the electric or magnetic field. Study of applications will also generate new algorithms as a consequence of the particular demands of a given application.

3.4.3 Worst-Case Bounds and Globally Convergent ML Algorithms for Parameter Estimation

Yoram Bresler, University of Illinois

Worst-Case Cramer-Rao bounds for parameter estimation

Another study of fundamental performance limitations involved bounds on the achievable accuracy in parameter estimation. These bounds, known by the name Cramer-Rao bounds, provide a benchmark for the performance of any estimation algorithm. No algorithm can achieve an accuracy better than predicted by these bounds. Under some condition, various estimators, most notably the maximum likelihood estimator, can achieve the performance predicted by these bounds. Although these bounds have been widely used in the sensor array processing literature, their application has had a major shortcoming: usually, they can only be evaluated numerically and used to generate plots for a particular scenario. The definition of this scenario for p emitters involves, in addition to the parameters of immediate interest (directions), also the specification of a covariance matrix with about p^2 free parameters. The latter so called nuisance parameters are usually not of interest. Owing to the large number of these nuisance parameters, it has been unfeasible to explore their effect in detail, and usually plots were generated for what were believed to be "typical" scenarios. Similar "typical" scenarios were used in numerical testing of algorithms. Unfortunately, there was no way to guarantee that the performance predicted for these "typical" scenarios is indicative of that for other scenarios.

To address this problem, we derived new *worst-case* Cramer-Rao bounds that eliminate the dependence on the amplitude nuisance parameters. These bounds provide both the worst and the best case bounds, thus determining a "performance envelope", which contains any actual scenario. Furthermore, this analysis also determined what are these worst and best signal scenario, so that they may be used in numerical and experimental studies of proposed algorithms and systems. The results show that the difference in performance between best and worst scenarios can be very significant, casting serious doubts on the aforementioned practice of studying what were thought to be "typical" scenarios. Instead, we believe that the new bounds should become the new "golden standard" for performance studies in sensor array processing, and more generally, in estimation problems involving superimposed signals, including radar, sonar, magnetic resonance spectroscopy, etc.

Algorithms for Maximum Likelihood parameter estimation of superimposed signals

The model of a signal consisting of superimposed signals of known parametric shape but unknown number and parameters arises in numerous applications. These include nonlinear regression problems in science and engineering, and in particular array processing, radar, sonar, geophysics, etc. Although conceptually simple, the fitting of the data with such a model involves extensive computation due to the nonlinear dependence of the likelihood function on the parameters. None of the previous methods could ensure global convergence to the global minimum of the multimodal likelihood function, except by infeasible exhaustive search. In this project we developed *Globally optimal* (the first of their kind) efficient estimation algorithms based on dynamic programming for the parameter estimation problem. One algorithm devleoped applies to a single experiment (snapshot) case, whereas another developed introduces some approximations further reducing the computational cost, and making it applicable to the multi-experiment case. A theoretical performance analysis of these algorithms confirms their merits, and predicts their limitations.

Imaging Applications As discussed earlier, using appropriate modeling, imaging can be reduced to parameter estimation problems. The results on fundamental limitations and algorithms can then be applied to these problems, reaping considerable benefits compared to classical approaches.

This idea was demonstrated in some detail in two techniques of image reconstruction from partial data. In the first approach, the scene is modeled as a combination of point targets and a relatively smooth background reflectivity map. Using a novel mixed parametric/nonparametric modeling formulation, optimal reconstructions of such scenes from limited Fourier data were obtained. The method improves considerably over the standard technique known as "clean" in radioastronomy, which produces many false targets due to mismodeling of diffuse objects such as nebulae by multiple point targets. Possible applications include radioastronomy and Radar.

The second technique is a suboptimum globally convergent and efficient algorithm for model-based restoration of an image modeled by piecewise-constant polygonal patches from its blurred (bandlimited) and noise corrupted version. The associated nonlinear optimization problem is solved by our dynamic programming algorithm for parameter estimation (see above). The technique has a broad range of applications, from astronomy and radar imaging, to electron microscopy, and provides superior performance over other deconvolution methods (including POCS) in the presence of high noise.

Algorithm for resolution of superimposed pulses of unknown and arbitrary shape

The separation and parameter estimation of superimposed pulses is a classical problem, arising in radar, sonar, communications, and geophysics. Normally, the pulse shape is assumed known, and only the number, positions and amplitudes of several echoes are to be determined. We considered the more challenging problem when the shape is also arbitrary and unknown. This is of interest in situations of a dispersive medium, extended targets of unknown reflective signatures, and operation of a covert passive radar. The algorithm developed for the solution of this problem exploits an approximate invariance structure in the frequency domain, which allows to apply the ESPRIT algorithm for parameter estimation. The new approach overcomes the bias of previous methods and their limitation to minimum-phase signals.

Application of Sensor Array Processing Techniques to Imaging

The work of this PI in the surveyed collection of projects ranges from fundamental bounds, through algorithms, to imaging applications. The common thread throughout is the overlapping signal model, which underlies sensor array processing, but is also fundamental to many other problems in science and engineering. As the field matures and better understanding of the fundamental limitation emerges, the focus is shifting to efficient computational algorithms, and ultimately to applications. While sensor arrays and the direction finding problem have inspired much of the research until recently, the rich body of knowledge and techniques is ripe for novel and creative applications and technology insertion in numerous other domains. To be sure, the research on communications applications reported by other researchers in this workshop will continue to be a major thrust area. However, the interest of this PI lies particularly in the imaging and image processing domain. In fact, often in these applications the models are more accurate and the assumptions easier to satisfy, than in the array scenario where propagation phenomena and miscalibration complicate the problem significantly. This PI (as well as other researchers) is now involved in intensive research into such imaging and other applications.

3.5 Towards Real-Time Implementation of Signal Subspace Algorithms

3.5.1 Sphericalized Subspace Updating

Ron DeGroat, University of Texas at Dallas

Introduction to Spherical Subspace (SS) Updating

- The ability to efficiently track the dominant eigencomponents or subspace of an estimated correlation matrix is an important part of many array processing (and some signal processing) applications.
- SS Updates are simplified rank-one eigen-updates in which subsets of the eigenvalues are preprocessed (i.e., forced to be equal) to generate spherical subspaces that can be deflated to reduce computation.If all of the eigencomponents of an $n \times n$ correlation matrix are tracked (without any sphericalization), the computational costs are $O(n^3)$ per update.The three most interesting cases of SS Eigen Updating are:
  - Signal Averaged (SA) Updating which tracks the Eigenvalue Decomposition (EVD) of a sphericalized correlation matrix with two eigenlevels, i.e.,

    $$R_{SA} = \begin{bmatrix} U_1 & U_2 \end{bmatrix} \begin{bmatrix} d_1 I_r & 0 \\ 0 & d_2 I_{n-r} \end{bmatrix} \begin{bmatrix} U_1^H \\ U_2^H \end{bmatrix}$$

    The SA Update has a computational complexity of O(nr) if only the dominant $n \times r$ "signal" subspace is tracked.
  - Signal Eigenstructure (SE) Updating tracks the dominant r "signal" eigencomponents and an average noise eigenlevel,

    $$R_{SE} = \begin{bmatrix} u_1 & \cdots & u_r & U_{r+1} \end{bmatrix}$$
    $$\times \begin{bmatrix} d_1 & 0 & 0 & 0 \\ 0 & \ddots & 0 & 0 \\ 0 & 0 & d_r & 0 \\ 0 & 0 & 0 & d_{r+1} I_{n-r} \end{bmatrix} \begin{bmatrix} u_1^H \\ \vdots \\ u_r^H \\ U_2^H \end{bmatrix}$$

    with a complexity of $O(nr^2)$.

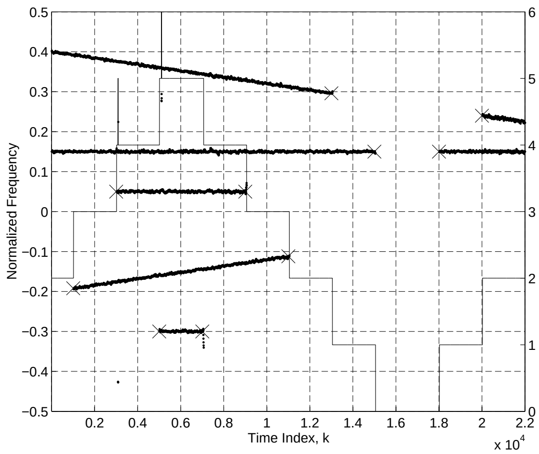


Figure 3.29: SA4 tracking scenario with nonstationary sources and changing signal rank. The starting and stopping frequencies are indicated with a large "X", SNR = 10 dB for all signals, n=10 and alpha=0.95. The frequencies are estimated with SA4 based root-MUSIC and the signal rank is estimated with a low pass filtered SA4–MDL signal rank estimator.

– Signal Averaged Updating with Four Eigen-levels (SA4),

$$R_{SA4} = \left[\begin{array}{cccc} U_1 & u_2 & u_3 & U_4 \end{array} \right]$$

$$\times \left[\begin{array}{cccc} d_1 I & 0 & 0 & 0 \\ 0 & d_2 & 0 & 0 \\ 0 & 0 & d_3 & 0 \\ 0 & 0 & 0 & d_4 I_{n-r} \end{array} \right] \left[\begin{array}{c} U_1^H \\ u_2^H \\ u_3^H \\ U_2^H \end{array} \right]$$

has a complexity of $O(nr)$. Like SE, the SA4 update converges quickly, but does so with the efficiency of SA. Unlike SA, the SA4 update can adaptively track the rank of the r-dimensional signal subspace by monitoring the boundary eigenvalues, d_2 and d_3, and chosing r so that d_3 is in the noise level while d_2 is not. See Figure for simulation with time varying rank and frequencies.

Basic Concept of Spherical Subspace (SS) Updating

SS Updating is a special case of Rank-One Eigen-Updating [Golub73, Bunch78]:

$$\begin{aligned} \tilde{R} &= \alpha R + (1-\alpha) x x^H \\ &= \alpha U D U^H + (1-\alpha) x x^H \\ &= U(\alpha D + \beta\beta^H) U^H, \qquad \beta = \sqrt{1-\alpha} U^H x \\ &= U(Q \tilde{D} Q^H) U^H \\ &= \tilde{U} \tilde{D} \tilde{U}^H \qquad \tilde{U} = UQ. \end{aligned}$$

BASIC IDEA OF RANK-ONE EIGEN-UPDATING:

- Find the EVD of $S = (\alpha D + \beta\beta^H) = Q \tilde{D} Q^H$
- Update the Eigenvectors: $\tilde{U} = UQ$

  NOTE: If any of the eigenvalues in D are repeated, the EVD of S can be deflated resulting in reduced computation [Bunch78].

  NOTE: Any EVD (or similar decomposition, e.g., the URV) method can be used to update the simplified matrix, S.

Summary of Current and Future SS Research

- Analysis of Sphericalized Subspace (SS) Dynamics: An ODE based Proof of Convergence shows that the sphericalized subspaces asymptotically converge to the true subspaces, in the mean.
- Forward/Backward SS (FB-SS) Updating: In many cases, improved subspace estimation performance can be achieved for less computation.
- SS Detection Schemes: With a Four Level SS Update, information theoretic criteria can be used to decide if the dimension of the Dominant (or Signal) Subspace should be increased, decreased or remain unchanged.
- Multi-Level SS Updating: any number of distinct eigenlevels can be used.
- Square Root SS Updating: The Singular Value Decomposition (SVD) update can also be sphericalized.
- Many different types of decompositions can be embedded within a sphericalized framework, e.g., the URV update.

- SS-RTLS: Sphericalized Subspace Recursive Total Least Squares.
- SS-RLS: Sphericalized Subspace Recursive Least Squares. SS-WRTLS
- SS-RLS and FB-SS-RLS adaptive filters that are based on updating a sphericalized version of the inverse correlation matrix are extremely stable and computationally efficient with convergence and misadjustments that are closer to RLS than LMS
- SS-Pencil Updating (for colored noise)
- Parallel and Fixed Point Implementation of SS Updating: Eigenlevels can be subjected to a saturation threshold without significantly affecting tracking capability.
- Improved Eigenvector (or Singular Vector) Stabilization Schemes, e.g., Cyclic Stabilization (where a different pair of eigenvectors is orthogonalized at each update) may be better than adjacent pairwise orthogonalization.
- Analysis of Time Varying Signals, Subspaces and Arrays
- New and Improved Algorithm Design based on Time Varying Analysis of Signals, Subspaces and Arrays

NOTE: Most of this work is/was done jointly by Ronald D. DeGroat, Eric M. Dowling and Darel A. Linebarger.

Summary of Related Research

- Experimental Microphone Array for Direction Finding and Beamforming
- Constrained MUSIC
- Constrained Beamspace MUSIC
- Linearly Constrained TLS
- Data Least Squares and other LS Problems
- Soft Constrained GSC
- TQR Adaptive SVD Updating
- Conjugate Gradient Eigenstructure Updating
- ML DOA Estimation in the Presence of Unknown Colored Noise
- Calibrated and Robust Adaptive DOA Estimation
- Computational Simplifications with the FB Data Matrix as well as the FB Correlation Matrix

Research Application Areas

- Underwater Direction Finding and Beamforming with Towed Array Sonar Systems
- Microphone Array Processing for Teleconferencing, Auditoriums, etc.
- Biomedical Pattern Recognition for Medical Diagnosis
- Adaptive Echo Cancellers and Channel Equalizers for Telecommunications
- Motion Estimation for Image Understanding and Analysis
- Rotationally Invariant Character Recognition
- Active Noise Control
- Mobile Cellular Telephones
- Speech Processing

3.5.2 Some Results in Algorithms and Architectures Development

Ray Liu, University of Maryland

OBJECTIVES and POTENTIAL IMPACT

Driven by the demands of high throughput and high computational complexity of modern signal processing, special-purpose and high-performance architectures are of great interest and of practical importance for applications such as adaptive array processing, automatic targeting, and high-speed communications. Rapid advances in VLSI/WSI microelectronics make it practical to build low-cost and high-density application-specific integrated circuits (ASIC) to meet the demands of speed and performance of modern signal processing. The scaling down of transistor sizes has made it possible to pack more than one million transistors on a chip. With this drastically increased on chip computing power, real time signal processing has come of age. While the dimension of the microelectronics technologies is reaching its ultimate limit, finding efficient parallel algorithms and VLSI architectures that may result in orders of magnitude of improvement in performance has become more and more important than before.

Recent developments of adaptive array processing relied heavily on efficient manipulation of linear-algebra-based algorithms. Not only the complexity and efficiency of the algorithms are of great concern, the numerical stability and parallel computation are also the key factors. The use of VLSI will allow the design and realization of application-specific structures and architectures for computationally intensive algorithms such as those used in linear-algebra-based signal processing or in efficient solutions of sparse matrices. In fact, as indicated in a recent report from the IEEE Circuits and Systems Workshop on Future Directions, *a growing number of important techniques are matrix-based and are historically more closely related to linear algebra than to signal processing. Many of these techniques are becoming increasingly important in signal processing and need to be blended with traditional algorithms in a compatible and complementary way.* A signal processing view-point brought to computational problems in linear algebra can potentially lead to new approaches to those problems.

We have witnessed the impact of these important cross-disciplinary research in the past decade, especially in the areas of adaptive array processing where sophisticated matrix-based algorithms constitute the heart of recent developments. The objective of this research is to develop efficient algorithms and architectures for real-time adaptive array processing. Many issues have been considered, and some of them are summarized as follows.

SUMMARY OF PROGRESS

URV-Based Subspace Algorithms and Architectures

Many problems in signal processing such as sensor array processing, spectral estimation, adaptive filtering, and image processing, require the computation of the rank of a matrix and an orthogonal basis for its signal space. The standard computational tool for solving such problems are either the singular value decomposition (SVD) or the eigendecomposition. If we think carefully we will find that in most signal processing applications, we really neither need to know the singular values (or eigenvalues) nor the singular vectors (or eigenvectors). What we need to know are simply the rank and an orthogonal basis that span the signal space. So, why bother to pay the computational price to compute the SVD? The answer to the past is that we have no choice given the numerical computational tools we have so far!

Recently, a new matrix decomposition, the rank-revealing URV decomposition, that requires much less computation than either a SVD or an eigendecomposition but reveals much of the same information has been introduced. The URV decomposition is in fact a new computational tool with properties in between the SVD and the QRD.

ESPRIT is an algorithm for determining the fixed directions of arrival of a set of narrowband signals at an array of sensors. Unfortunately, its computational burden makes it unsuitable for real time processing of signals with time-varying directions of arrival. We developed a new implementation of ESPRIT that has potential for real time processing. It is based on a rank-revealing URV decomposition, rather than the eigendecomposition or singular value decomposition used in previous ESPRIT algorithms. We demonstrate its performance on simulated data representing both constant and time-varying signals. We find that the URV-based ESPRIT algorithm is effective for estimating time-varying directions-of-arrival at considerable computational savings over the SVD-based algorithm.

We also consider a parallel architecture for updating the URV decomposition on a wavefront array. The wavefront array provides an efficient real-time mechanism for adaptive computation of the null space of a matrix as well as for handling rank changes during updating. Our emphasis is to develop an architecture that can implement both URV and QR decomposition so that all computational issues related to adaptive array processing can be performed on one single parallel processing environment.

Square-Root and Division Free Algorithms and Architectures

Efficient implementations of the recursive least squares (RLS) algorithms and the constrained recursive least squares (CRLS) algorithms based on the QR decomposition (QRD) have been essential for many adaptive beamforming and target tracking algorithms. It has been shown that the QRD-based algorithms have good numerical properties. However, they are not very appropriate for VLSI implementation, because of the square root and the division operations that are involved in the Givens rotation and the back-substitution required for the case of weight extraction.

Up to now, the planar (Givens) rotations are the most commonly used methods in performing the QR decomposition (QRD). But the generic formula for these rotations requires explicit square-root and division computations, which are quite undesirable from the practical VLSI circuit design point of view.

In this project, we discover a class of square-root-free family and a class of square-root and division free family to perform the QRD and RLS computation. We established the algebraic relations for the existing and new computational algorithms. We choose a specific instance for each one of the two parametric algorithms and make a comparative study of the systolic structures based on these two instances, as well as the standard Givens rotation. We also consider the architectures for both the optimal residual computation and the optimal weight vector extraction.

The dynamic range of the newly proposed algorithm for QRD-RLS optimal residual computation and the wordlength lower bounds that guarantee no overflow are presented. The numerical stability of the algorithm is also considered. A number of obscure points relevant to the realization of the QRD-RLS and the QRD-CRLS algorithms are clarified. Some systolic structures that are described in this paper are very promising, since they require less computational complexity (in various aspects) than the structures known to date and they make the VLSI implementation easier.

3.6 Experimental Array Systems

3.6.1 Experimental Sensor Array Systems for Mobile Communications and Semiconductor Manufacturing

Guanghan Xu, University of Texas at Austin

Experimental Antenna Array Systems for Mobile Communications

Objectives

The objective of this research effort is to implement and validate advanced array signal processing techniques (or smart antenna technology) to significantly expand the channel capacity, improve the quality, and reduce the cost of various wireless communication systems.

Potential Impact

The demand of wireless communications is growing exponentially during the last five years and it is conservatively projected that by the year 2000 the number of users will rise up to 115 million nationwide. With such rapid growth, it is obvious that current cellular technology will be incapable of handling sufficient numbers of cellular phone calls simultaneously, because the spectrum allocated for mobile communications is limited. The smart antenna technology is the enabling technology to make significant expansion of channel capacity so as to accommodate the growing demand without requiring more bandwidth. Furthermore, a smart antenna system at a base station, can also significantly improve the quality of services, increase the coverage, and reduce the cost of the RF front end. Finally, due to powerful receiving capability of the smart antennas, the cost of handset can be significantly reduced and its battery life can be considerably increased. The ultimate impact to the society is that more and more people can enjoy the convenience of wireless communications at reduced cost.

Progress to Date and Milestones

After 1-year hard work, we have completed a flexible and advanced smart antenna testbed that enables us to conduct various channel propagation studies and more importantly, to validate and demonstrate the advanced signal processing techniques we developed.

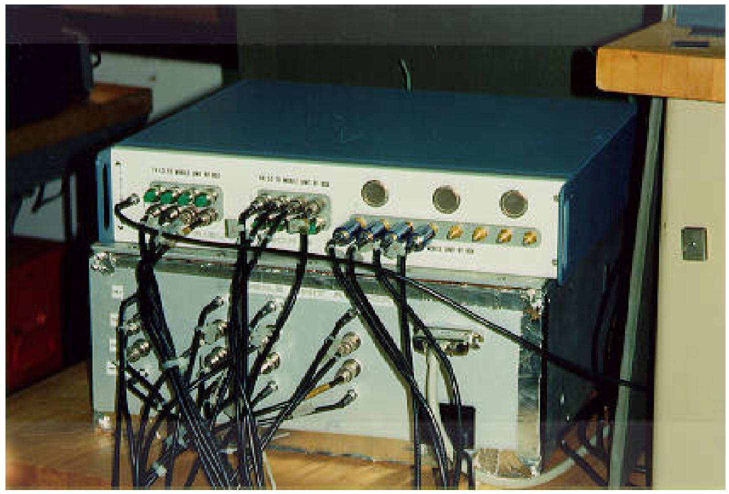
(1) one 10-element patch antenna array and one 8-element monopole antenna array; (2) 12 RF and IF downconverters and upconverters and switches; (3) two distribution boxes providing synthesized sources for RF and IF converters; (4) 12 A/D's and 24 D/A's; (5) 4 digital multiplexing and demultiplexing boards; (6) high-speed I/O boards and Sparc 10 console.

The unique capabilities of our testbed allow us to conduct various experiments related to different forms of wireless communications including cellular telephony, wireless LAN's, PCS, and LEO satellites. Our flexible RF fronts, A/D's, D/A's also allow us to emulate different multiplexing schemes including frequency-division-multiple-access (FDMA), time-division-multiple-access (TDMA), and code-division-multiple-access (CDMA).

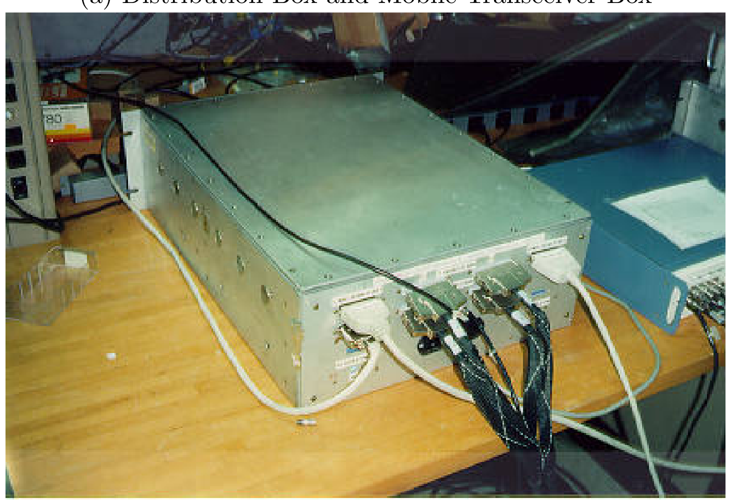
(a) Base-station Antenna Array

(b) Mobile Antenna

Figure 3.30: Experimental smart antenna array system developed at UT Austin.

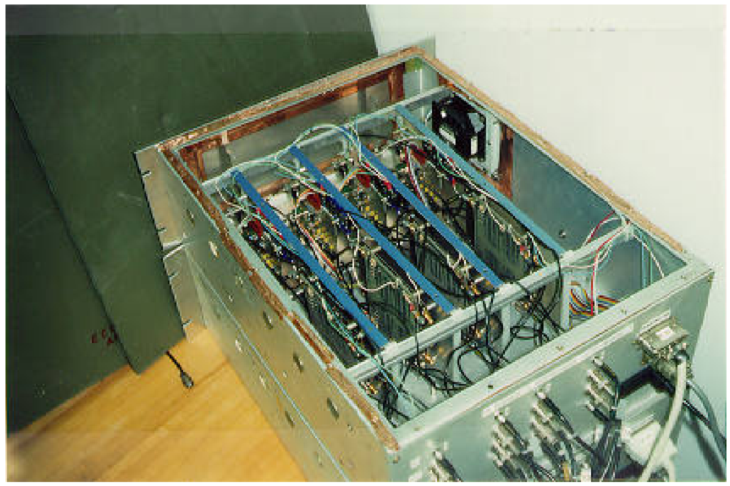


(a) Distribution Box and Mobile Transceiver Box

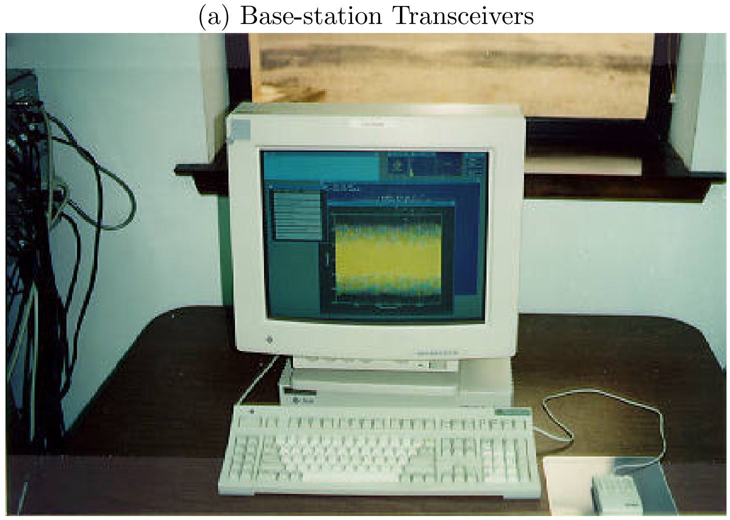


(b) Multiplexer and Demultiplexer Box

Figure 3.31: Components of experimental smart antenna array system developed at UT Austin.



(a) Base-station Transceivers



(b) Sparc 10 Console

Figure 3.32: Components of and computer interface to experimental smart antenna developed at UT Austin.

In short, this is the first advanced smart antenna testbed among all the academic institutions nationwide.

Experimental Sensor Array System for Semiconductor Manufacturing

Objective

The objective of this research effort is to develop acoustics based approach to obtain on-line measurement of wafer temperature profile in a rapid thermal processing chamber for semiconductor manufacturing.

Potential Impact

Rapid thermal processing (RTP) is a state-of-the-art technique for performing the necessary wafer fabrication operations of annealing, oxidation chemical vapor deposition, and other semiconductor manufacturing processes, in a single chamber within orders of magnitude shorter period of time. Although the RTP process is the future of semiconductor manufacturing, one of the most serious problems that hinder the commercialization of this technology is the difficulty of tightly controlling the wafer temperature profile during the RTP process. This difficulty stems from the inability of most existing techniques to measure wafer temperature profile accurately and rapidly. If successful, this new approach based on both acoustic sensor array hardware and advanced signal processing techniques can solve this serious problem and significantly speed up the process of commercialization of the RTP technology.

Progress to Date and Milestones

During last six months, our research and development effort has been focused on the two areas: algorithm development and testbed development.

We have conducted extensive investigation about the spread spectrum approach and development of new signal processing techniques for increasing the measurement rate. Though look promising in theory, the spread spectrum approach is not feasible since the signal rate is too high to be feasible. However, we have managed to develop a multi-tone approach to achieve the same goal, i.e., significantly increasing the measurement rate.

We have also spent much time determining the requirements and functional specifications of the temperature measurement testbed and have actually designed the testbed. So far we have completed the quartz and transducer part of our testbed and are currently testing the completed part. Development of the electronic part of the testbed is underway.

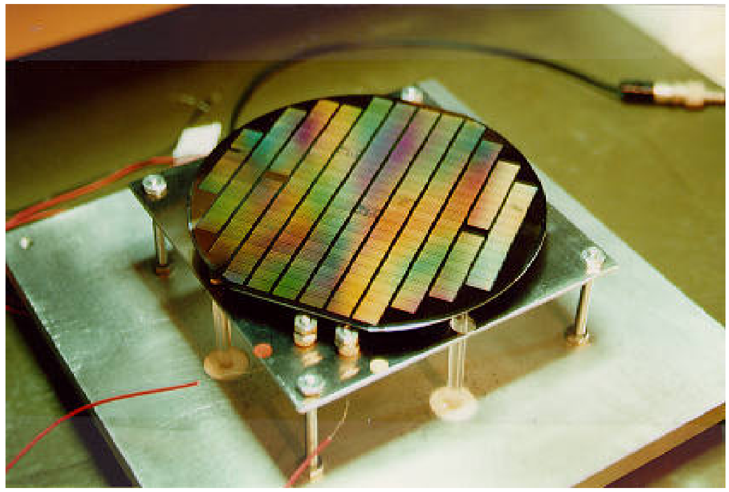
3.6.2 An Experimental Program in RF Environment Characterization for SDMA-Based Wireless Communication Systems

Richard Roy, ArrayCom, Inc.

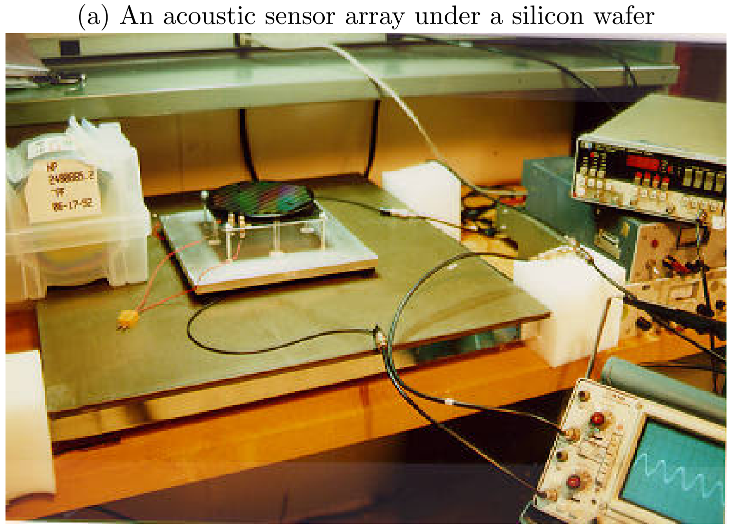
Introduction

Wireless communication systems are an increasingly pervasive method of information transmission throughout the world. As radio spectrum is a finite resource, the success and the growth potential of such systems are critically dependent upon its efficient use. The capacity of some systems has already been exhausted, for example cellular telephone systems operating in certain urban areas. Spectral efficiency is typically sought by employing modulation schemes such as Code Division Multiple Access (CDMA), Time Division Multiple Access (TDMA), and Frequency Division Multiple Access (FDMA). Efficient modulation is an important component of a well-designed wireless system, but it fails to alleviate what is perhaps the most important and common spectral inefficiency present — spatial inefficiency.

This inefficiency can be significantly mitigated by incorporating directional transmission and reception capabilities into wireless systems. With directional communication, multiple conversations can be supported on a single channel as the majority of the radio energy associated with each conversation is transmitted directionally between the two endpoints. Directional communication also reduces transmitted power requirements by obviating the need for omnidirectional coverage. Reduced transmitter powers, in turn, result in reduced levels of background radio frequency pollution, and therefore channels can be reused more frequently in space. By supporting multiple conversations on a single channel and allowing increased spatial reuse of channels, spatially directive transmission assuredly offers significant capacity increases over current systems. Spatially directive communication, utilizing arrays of antennas and sophisticated digital signal processing techniques,



(a) An acoustic sensor array under a silicon wafer



(b) Acoustic Temperature Measurement System for Semiconductor Manufacturing

Figure 3.33: Experimental smart acoustic array system developed at UT Austin for on-line measurement of wafer temperature profile in a rapid thermal processing chamber for semiconductor manufacturing.

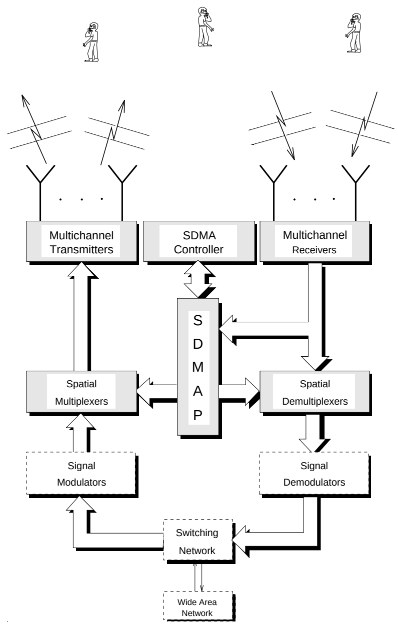


Figure 3.34: Canonical wireless system base station employing SDMA

is referred to as Spatial Division Multiple Access or SDMA — an area pioneered by ArrayComm.

A canonical wireless system base station employing SDMA is depicted in Figure 3.34. Conventional RF receivers are used to suply signals to the SDMA processor (SDMAP) and the spatial demultiplexers. Within the SDMAP, the antenna data are processed to determine the number of users present on the channel, to track users, and to generate the information necessary to demultiplex the co-channel signals. Using this information, the spatial demultiplexer separates, demodulates and passes the users' signals on to the remainder of the network. Complementary operations are performed on transmission. Information provided by the SDMAP allows the spatial multiplexer to combine signals on a single frequency channel so that each user receives their intended signal without interference from the other users' signals. Multiple frequency channels are supported by replicating the basic SDMA functionality for each channel.

Objectives

The effort being conducted under this program involves completion of a fully functional real-time SDMA prototype for performing experiments to characterize typical wireless communications radio frequency (RF) environments experienced by arrays of antennas. The results of the experimentation will be used to refine the algorithmic components of SDMA. Since real-time implementation (and cost) are driving factors in commercialization of this technology, significant emphasis will be placed on these factors in the process.

In particular, the effort has three specific objectives, as described below.

1. **Construction of a full-featured SDMA prototype**
   A flexible, fully functional SDMA prototype system will used. The prototype system will contain hardware and software that implements the SDMA strategy (namely bearing estimation, tracking, spatially selective transmission, and spatially selective reception) in real-time, along with the capability of storing data for off-line analysis where practical. The algorithmic components of SDMA will be implemented digitally at baseband. Digital modulation formats will be incorporated where possible, to allow for testing of recently developed multidimensional signal structure-based algorithms as well as the development of new techniques for exploitation of known signal structure in such systems.

2. **SDMA algorithm refinement and development**
   The algorithmic portion of the effort will be twofold. Significant effort will be directed at making the algorithmic components work in concert at real-time rates (*i.e.* spatial channel update rates on the order of several Hz). Further algorithmic effort will be directed at the issue of tracking (*i.e.* assuring that the spatial channel for each handset is reliably updated on the basis of the handset's position). Finally, research into and development of algorithm modifications necessary to ensure reliable system performance in the presence of a wide variety of hardware errors will be undertaken, and experiments will be conducted with the candidate algorithms to verify their performance.

3. **Characterization of the SDMA RF environ-**

ment
Existing studies of typical wireless communication RF environments are not particularly useful for SDMA algorithm development and performance evaluations. In those studies, measurements are taken almost exclusively with a single antenna. To be useful for SDMA applications, measurements must be taken with an array such as is used in the SDMA system. Once the prototype has been built, the entire system will be transported to several different locations (a portable extensible antenna mast has already been secured by ArrayComm) and measurements will be taken to characterize representative SDMA RF environments in rural, suburban and urban areas. The results of these test will provide the critical information necessary for algorithm refinement, robustification, and product development.

In summary, this effort is directed toward obtaining a sufficient amount of experimental data under real operating conditions to allow an accurate assessment of the viability of SDMA technology in real-world environments. The experience and results of these experiments will be crucial to the development of reliable, cost-effective, efficient SDMA algorithms, and ultimately products for addressing the ever-increasing demand for wireless access in today's communications marketplace.

Technological Challenges

The technological challenges facing this effort can be roughly divided into two categories, practical and theoretical. From a practical standpoint, some of the relevant issues that will be addressed in this effort include:

- downlink data collection
- antenna element design (mutual coupling)
- Rx linearity and dynamic range
- Rx bandwidth and ADC bits/rate cost tradeoffs
- Tx PA linearity and power cost tradeoffs
- Tx bandwidth and DAC bits/rate cost tradeoffs

To assess the performance of any SDMA-based full-duplex mobile wireless communication system, the issue of downlink data collection must be addressed. The basic problem is that assessment of the performance of any strategy for differential spatial distribution of energy seems to require that the spatial distribution be measured. This is certainly a non-trivial task, especially considering the underlying nonstationarity of the environments in which the tests will be conducted. The remaining issues are of a practical engineering nature and relate directly to cost/benefit trade-offs with significant impact on final productization decisions. While they are not on the list of burning theoretical issues of our time, they are certainly the most important of the practical issues that will determine the ultimate cost/benefit trade-offs facing manufacturers and operators of SDMA-based product and services.

At the "boundary" between practical and theoretical, there are a number of interesting challenges that will arise, including:

- antenna array design (f(environment))
- "micro" versus "macro" arrays
- the range of RF environments and how to model them
  $$\textit{stochastic} \longleftrightarrow \textit{deterministic}$$
- algorithms as a function of environment ... what can be exploited?
- real-time implementation of such algorithms
- system design issues - how many antennas ... where

From the experimental effort, a characterization of various representative RF environments (at a particular frequency) is expected. Of interest is to assess whether it is possible to come up with a single parametric model of the RF environment that, with appropriate identification of the parameters, can sufficiently accurately characterizes these environments. This unification could lead to the development of a generic class of algorithms for addressing these environments in a uniform way. Currently proposed solutions are heavily dependent upon the environment. Solutions proposed for harsh urban environments do not provide the benefits possible in the less complex suburban and rural environments, and those proposed for the less complex RF environments do not perform well at all in more complex environments.

From a purely theoretical standpoint, the following are interesting challenges facing researchers in this area (not all of these will be addressed in this effort):

- air interface protocol design - feedback or feedforward (intelligent or blind)
- network protocol design for spatial processing - centralized versus decentralized intelligence in the network
- capacity implications of both
  - what is the ultimate capacity measured in bits/sec/Hz/volume of such systems?
  - what is the size of a bit?
- "wired" versus "wireless" information flows in such networks

From a "next generation" SDMA system implementation perspective, the issue of air-interface protocol design is important. Taking into account that base stations are "smart", how should protocols be designed to make maximum use of this feature? If "more intelligent" subscriber units are allowed, how much intelligence should they possess? What are the network level impacts of SDMA technology and how should network protocols be designed to exploit intelligent air-interfaces? Ultimately, the question that will need to be addressed relates to the fundamental issue of capacity. In a general form, the question could be:

> *In a given volume of space, what is the maximum information flow between two arbitrary sets of sources and sinks?*

While this may seem overly simplified, the complexity quickly surfaces when one considers (relativistic) space-time issues and the physics of information (RF) propagation.

Applications of SDMA Technology

It is widely recognized that the worldwide market potential for wireless communication systems significant, and the demand is great. Wireless local loops are the local phone systems of the future for developing countries as well as eastern Europe. Two-way electronic messaging, PCN's, and next generation cellular systems are under development. These markets are expected to reach over $100B annually by the year 2000 worldwide.

These and other similar systems all face a critical problem of scarcity of the fundamental resource — electromagnetic spectrum. SDMA has the potential to provide substantial relief from this problem. Once developed, SDMA-based products would certainly have large market potential in a wide variety of marketplaces worldwide.

3.6.3 Adaptive Arrays for Wireless Communication Systems With Multipath

Jack Winters, AT&T Bell Laboratories

Objectives

The goal of this research is to derive techniques for, analyze, and implement adaptive arrays in wireless communication systems with multipath.

Progress to Date

We have studied signal processing techniques for increasing the capacity and reducing signal distortion in fiber optic, mobile radio, and indoor radio systems. We are currently studying adaptive arrays and equalization for indoor and mobile radio.

Major Research/Technology Problems.

The major problem is demonstrating adaptive arrays in operating wireless systems and developing cost-effective implementation techniques for commercial systems.

3.6.4 Tradeoffs Among Hardware, Software and Algorithms so That a "Small" Group can Build a Large Array

Harvey Silverman, Brown University

Introductory Remarks

Many modern systems are now based on the application of complex mathematical algorithms using advanced computing implementations. Most of yesterday's algorithmic research is making its way into products not only used in esoteric military/strategic applications but also by the general consumer. Today's technology, however, has also had an important impact on today's algorithmic research; it has become essential to test proposed algorithmic advances on real data and in real-time implementations.

Consider the algorithm research and development process.

- **Stage I:** Gain insight into a need for and algorithm or an improvement on an existing algorithm.
- **Stage II:** Create the kernel of the algorithm – typically mathematically on paper.
- **Stage III:** Implement on the computer and test on simulated data.
- **Stage IV:** Compare to existing algorithms using simulated data. Repeat stages II and III and IV until it works well.
- **Stage V:** Obtain and run on real data (offline). Repeat stages II - V until it works well.
- **Stage VI:** Implement in real-time environment, i.e., develop hardware, software and variations of the algorithm until it works.

As a rule, until the early 1970's when the size of RAM for large computers grew to the order of megabytes, the development of most algorithms was impeded by not being able to perform meaningful simulations in Stage III. Only very few places, such as IBM Research, AT&T Bell Laboratories..., were able to work on real data in the 1970's at great expense. At that time, even these large organizations did not do much to implement their algorithm developments in real time.

Although minicomputers and larger time-shared computers helped for developing some algorithms in the late 1970's and early 1980's, these machines had too high a cost (complexity) for gathering real data and too little power to implement substantive experiments. Not much really changed until the advent of the higher-performance desktop workstation and/or the higher-end PC of about 1990. Not only was the cost of computation reduced to under $500 per MIP, but both local RAM and hard disk storage was now available in sufficient quantity to allow the obtaining of real-data. Off-the-shelf A/D and D/A interfaces to the real-data world became widespread at about this time. A new era began for those who were developing computational algorithms; stages III, IV and V, at least, could now be run everywhere. Simpler algorithms made for relatively low sampling rates could even be run in real time on these conventional machines.

Most research algorithms still test the limits of computational power. However today's technology that allows the offline experimentation on real data using conventional hardware also allows special-purpose hardware to be built that can compute most algorithms in real time. Thus, for the first time, perhaps, stage VI work is feasible for many more research groups. Not only can the problem of making the algorithm work comparatively well on real data offline be addressed, but also some of the insideous problems due to implementational constraints, variable environments, and an infinite data stream may be brought under the research umbrella. One may easily predict that the technology will grow quickly to support having real-time capability with off-the-shelf hardware for many more algorithm-exploration research areas. Thus it seems that the rubric of research that ends in stage III will have less and less impact relative to those who take their algorithms through all six of the stages above.

Growth in I-VI Research will not only be due to the advances in technology, but also due to the self-interest of the researcher as the funding scenario changes. As more research is able to be brought out as hardware/software or simply software products, universities, in line with the current practice in industry, will make strong efforts to grow patent portfolios. The practice of licensing from the universities, today a very small part of the research funding scene, will grow markedly in importance both because of the self-interest of the inventors and universities and the likely diminution of the sources of Federal funding. I-VI research is essential for the patent process and to have sufficient **important and protectable** intellectual property for a potential industrial licensee to negotiate an arrangement with the inventors and university. Thus, aside from the rare brilliant inspiration, it seems that the real impact will stem only from I-VI Research. As a consequence, it will be evident that the payoff for I-III, I-IV or I-V Research will generally be smaller than I-VI efforts, and thus general support for the former activities will diminish.

The Presentation of H. Silverman (4-28-95)

Relative to Architecture and Construction
of "A Large-Scale, Intelligent Three-Dimensional
Microphone-Array Sound-Capture system"

James F. Flanagan and Harvey F. Silverman PI's

The purpose of this presentation was to present some of the hard-learned lessons gained from building large-scale computational and data acquisition support for the real-time implementation of algorithms. A set of general ideas, very presumptuously labeled **axioms**, was presented first. Then, some of the principles were shown in the light of an example, our large microphone-array construction project, called the Humongous Microphone Array (HMA). The following is the setting for the **axioms**:

- **Real time**
- Large number of sensors $\Rightarrow$ high I/O requirements
- 'High' computational capability
- 'Low' latency-time a factor
- Fairly general purpose hardware/software system
- Easy to program and change algorithms
- Reasonable physical size and electrical power requirements
- **Really Completable!!**

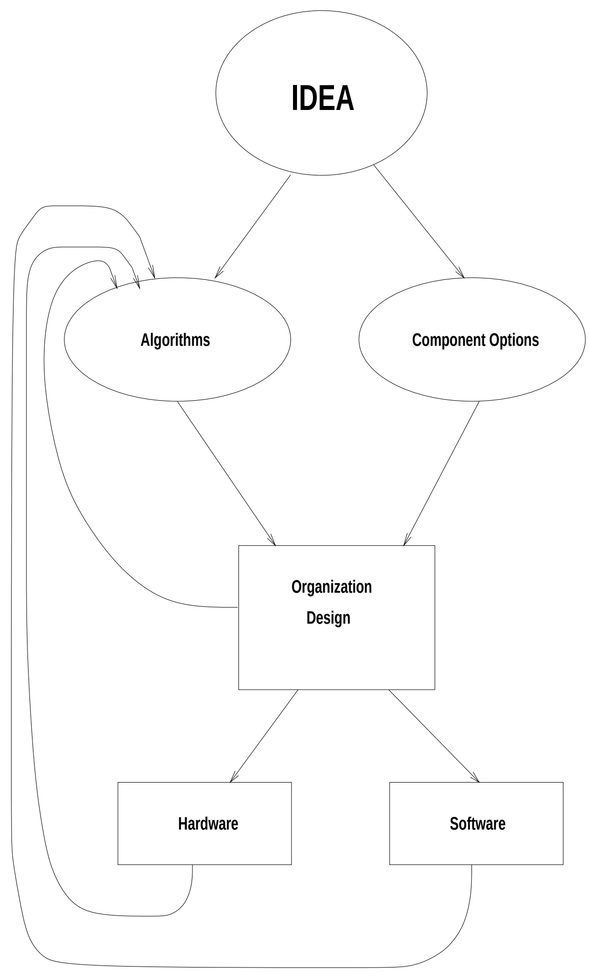


Figure 3.35: Design of a Large, Real-Time System

The design of a large real-time system is an iterative one. This is illustrated in Figure 3.35. It is most important to realize up front that the easiest part of the system to change is the algorithm! Thus, the design and construction of a specialized, large real-time system must necessarily include the algorithm developers who should be willing to modify the algorithm to allow the hardware/software components to be developed more quickly and with lower cost. Just "throwing the algorithm over the transom" will result in a more complex and more expensive design that will also take longer to build.

It is not easy for a small group to complete a large construction project at a university. Our definition of a small group at Brown is 1) 3 months of PI time (1/4 of all efforts), 2) 1/3 of the time of a super analog/digital design engineer, 3) 1/4 time of a good technician, 4) 4 PhD students adding up to about 2.0 full time PhD students who are working on software and algorithms. The effort levels at Rutgers are about the same.

The following list of **axioms** represents some guidelines for building a large project that have been gleaned from experience:

1. Develop and test a set of computable, representative algorithms **first**.
2. Select silicon (microprocessors, memory A/D, etc.) that you can **get**.
   - **Corollary:** On NSF Projects, get it for **FREE**.
3. Use "off-the-shelf" systems where feasible – in particular, buses, I/O and workstations.
4. Design to use the smallest number of reasonably-sized, different PC boards as possible.
5. Design hardware to make the development of an operating system, and subsequent user programming, easy.
6. Keep the above axioms, but do not "throttle" I/O.
7. Develop designs that have 'side' benefits.
8. If it does not quite work – adapt the algorithms!!

It is this set of axioms that have governed the HMA project. The HMA is to be a real-time system that supports localization of sources and beamforming in real time for 512 microphones. It is meant as a research system to uncover the effectiveness and the real issues of using such and arrray in small and large rooms for many applications such as teleconferencing, speech recognition and theater. An overview of the architecture is shown in Figure 3.36. In addition to doing location and conventional beamforming, several other algorithms, such as matched filtering to a number of reverberations in addition to the direct source, are intended to be implementable in real time. The architecture of the system has been developed to make the development of an operating system with arbitration unnecessary. There are two major projects – each a duplicatable printed-circuit board – in the project. (There are two more single-shot boards, but these are relatively simple.)

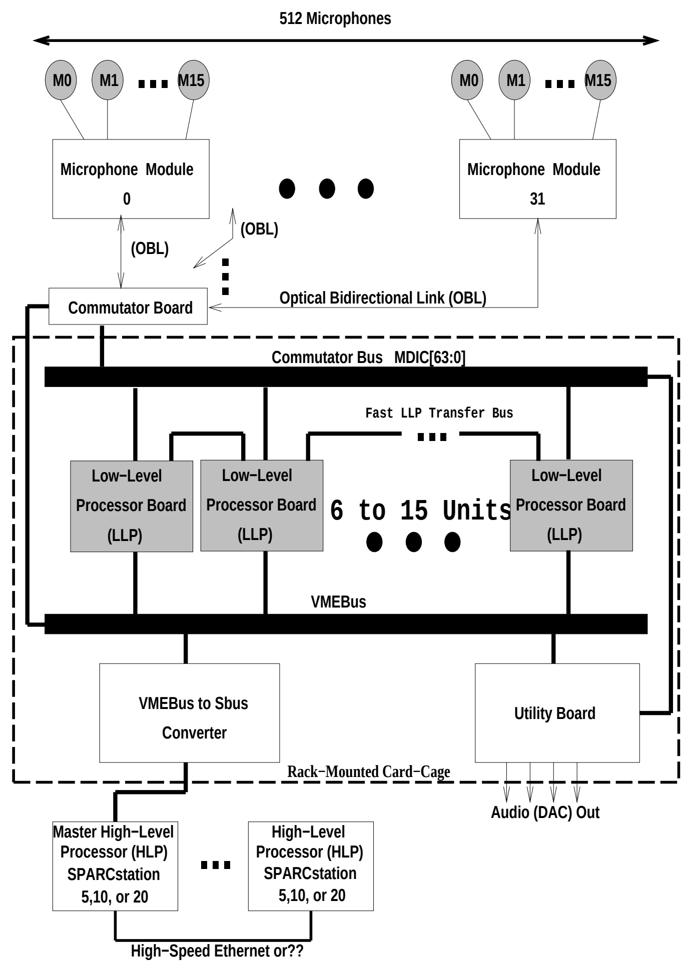


Figure 3.36: The HMA Architecture

The first board, called the *module* board, being designed at Brown, is shown in Figure 3.37. It converts the signals from 16 microphones, takes them to the frequency domain and sends the data along a fiber-optic channel to the console of the system. Each module board has a single 33MHz AD21020 DSP microprocessor, suitable fast memory, A/D's etc. Thirty-two of these boards are to be used in the system. Currently, the first printed-circuit versions of the boards are being debugged. Obeying one of the axioms, each of these boards has a second potential application as a stand-alone processor through an interface to a personal computer. There are several boards that we are to make for some potential users. VTEL, the video conferencing company in Austin, TX not only will be using the board in this configuration, but also helped in the board layout. It is likely that additional systems will be built for ARPA/FBI and UCLA. Each module board in the HMA, communicating via fiber optic cable, could be as much as 3Km from the console!

The 32 module signals are combined at the console and put on a one-way backplane bus 64 bits wide and operating at about a 20MHz frequency. This bus is used as input for every low-level-processing board. These boards, being designed at Rutgers, each will have about eight ADSP21020 microprocessors, hooked up so that no arbitration need be used in the operating-system software. Rather, another processor on the board, likely from the 680X0 family, is used to schedule I/O by taking advantage of the two-bus architecture of the DSP chips. By this mechanism, board-level memory buffers may be loaded simultaneously with computation, and I/O to the individual processors may be done through the simple HALT mechanism. Another advantage is that processors doing the same task, e.g., some DSP for a specific-sized subset of the microphones, run exactly the same programs. All processors are loaded over a standard VME bus through an S-Bus to VME-bus converter from a SUN SPARCstation 5 workstation. It is planned to have about 15 low-level processor boards, implying the power of some 120 DSP systems!

The ideas of the software design are made clear in the following list:

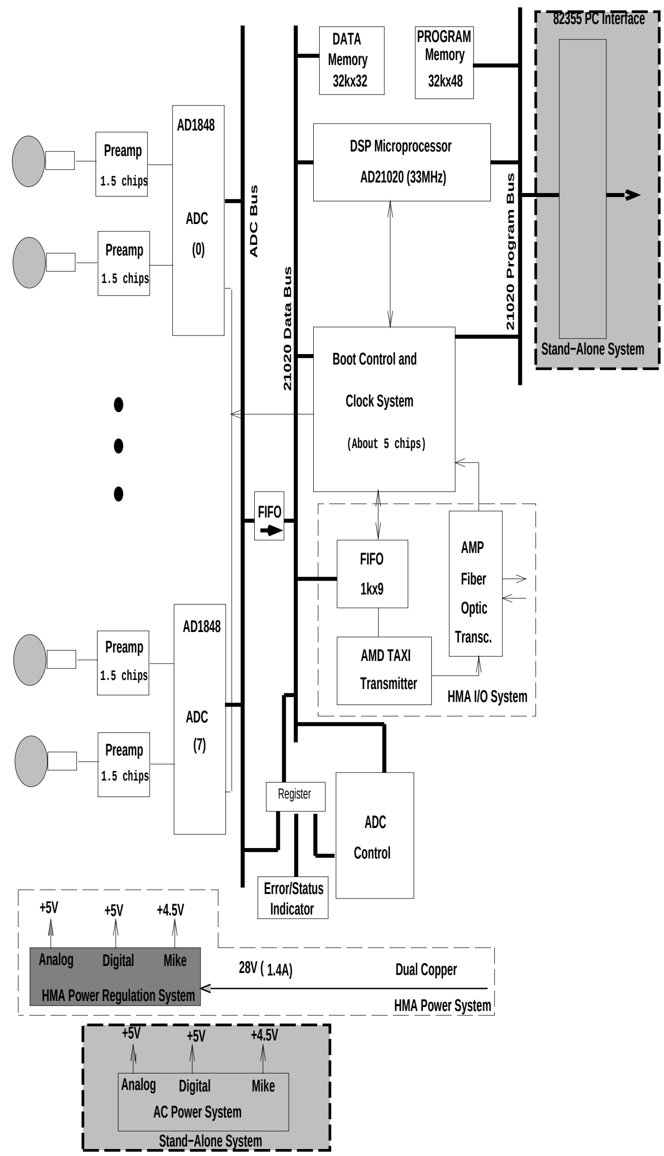


Figure 3.37: The HMA Module Board

- PHILOSOPHY
  1. **Keep it simple!!!**
  2. **NO** conventional operating system

3. Data never influences program flow ⇒ it can be done

- Load and Go System for the User
  - Assigns programs to processors - booting
  - Builds routing tables fore data
  - Inserts parameters – e.g., microphone positions, number of mics actually used etc.
  - Checks for errors – mismatched data types, too many mics assigned, etc.
  - GUI for input and data flow visualization
  - A talented undergraduate can build it on a standard workstation

Once again, the HMA system may be seen to be being built using the axioms as pretty hard-and-fast guidelines.

One may further see the design process as depicted in Figure 3.35 as applied to the HMA by looking at Figure 3.38. This figures shows, in block-diagram form, the flow of one of the algorithms that will certainly be run on the HMA. The important interactions among algorithms, data flow, processors and control software are evident.

The status of the HMA project as of June 21, 1995 is:

- Hardware
  - Console support complete
  - First 3 of 6 module boards working
  - Commutator board in debug and utility board designed
  - LLP design in process

- Software
  - Module software complete
  - Load and Go system in process
  - Brandstein application in process

In summary, the HMA project is a very large building effort, at least for a university, and requires a lot of cooperation from industry. Donations of semiconductor components from their makers, as well as the donation

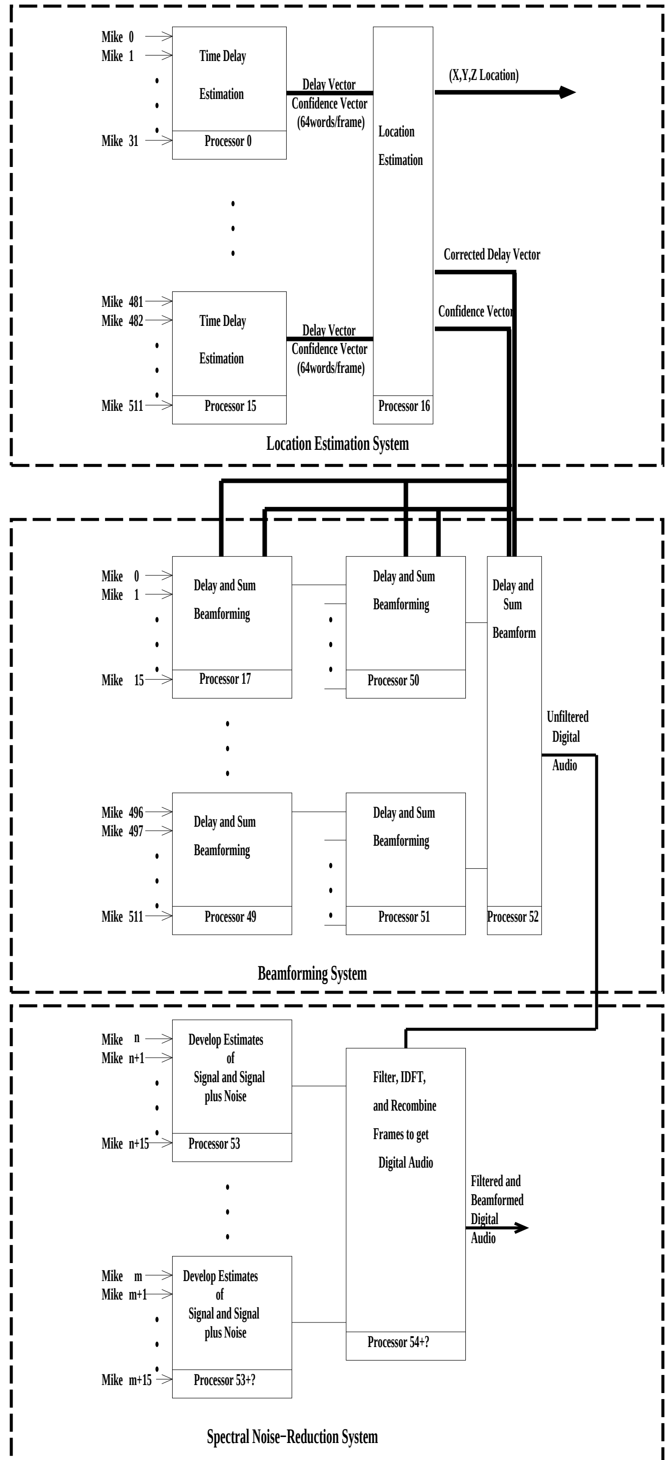


Figure 3.38: Brandstein Location, Beamforming on the HMA

of many other components is an essential aspect of the task. This part of the project requires a large and unexpected piece of time and effort. Further advantage has been gained from the side-benefit application, as one of the major board-layout efforts was done by an expert from industry.

It is key that the HMA project has obeyed the axioms presented. Even though the project is still a major "push" for those involved, the probability of success would have been close to zero were not the axioms applied. In any event, the project is an engineer's delight and all involved have great excitement.

3.7 "Array" of Applications

3.7.1 Novel Algorithms for Finding Localized Energy Solutions With Application to Magnetoencephalography (MEG)

Bhaskar Rao, University of California at San Diego

Introduction

Medical imaging has been and stills remain an area of active interest with array processing playing an important role in the reconstruction of an image from measurements. Of particular interest, and of main concern in this research, is the high resolution imaging of the brain with the use of these images being to obtain a better understanding of the functioning of the brain, and to provide effective tools for clinical diagnosis. Some key requirements on the technology for this goal to be realized are

- High temporal resolution. Imaging the activity in the brain requires tracking electrical activity that last one to several tens of milliseconds. Imaging methods like single-photon-emission computed tomography (SPECT) and echo-planar based functional magnetic resonance imaging (MRI) though useful do not provide such high temporal resolution (typically the resolution is in secs).
- High Spatial resolution. Since activity in the brain is localized, both for understanding and diagnosis purposes high spatial resolution in the reconstruction is desired.
- Noninvasive. The methods for imaging have to be noninvasive. This makes the procedures safe, psychologically more acceptable, useful for early diagnosis and for practicing preventive medicine.
- Cost effectiveness. Medical costs are a very important element in the introduction of new technology. Keeping the costs for treatment down is an important consideration.

A candidate that shows considerable promise is Magnetoencephalography (MEG)[1,2]. In MEG, magnetic field measurements are made on the surface of the head using an array of sensors. The sensors in this case being Superconducting QUantum Interference Device (SQUID) magnetometers. The magnetic field measured is a result of source currents in the brain that result from neuronal activity. Reconstruction of these source currents from the magnetic measurements results in an image of activity in the brain which can be used for understanding and diagnosis purposes. MEG appears to be able to address all the issues mentioned above. It is noninvasive, and is capable of producing high temporal and spatial resolution. With the improvement of instrumentation, cost for these systems are projected to go down and coupled with reduced hospitalization costs become cost effective. The technology also is promising for imaging other organs, and for non-destructive testing.

There is considerable work that needs to be done both on the instrumentation and signal processing side before the technology is commercially viable. Our research addresses array signal processing issues in MEG. The processing goal in MEG is the accurate reconstruction of source currents representing neuronal activity and essentially involves dealing with the electromagnetic inverse problem. Solving this inverse problem poses many challenges which we discuss next, and the main objective of our work is to develop novel algorithms to solve such inverse problems.

Technical Challenges

The issues that arise in the MEG array processing problem are challenging and have commonality with other array processing problems. It involves modeling issues,

[1] J. P. Wikswo, IEEE Trans. on Applied Superconductivity, June 1995

[2] M. Hamalalainen et al, Rev. Mod. Phys., April 1993

dealing with highly dynamic environment (nonstationary), low signal to noise environment, limited measurements, and the need for high resolution reconstructions. Some of these problems are discussed next.

On the modeling front, an issue of interest in this research is the modeling of the current sources that represent the neuronal activity. A popular approach is the use of multiple dipole model to describe the current sources. This reduces the inverse problem to a parameter fitting problem. Popular parametric methods can then be adapted to solve the source localization problem. Alternative approaches, which do not make the few pointlike sources assumption, employ a nonparametric formulation and attempt to find more general solutions to the inverse problem. How to best obtain localized reconstructions using nonparametric methods, and how to synergistically combine parametric and nonparametric methods for MEG reconstructions is an important issue. Also devising effective procedures to account for the anatomical shape and parameters in the forward model that arises in the inverse problem is important.

Since the goal of MEG is brain mapping, we are dealing with a highly dynamic environment. Though multiple snapshots are available, extensive averaging over snapshots limits time resolution. How to deal reliably with multiple snapshots for both high temporal and spatial resolution is an important issue. Tradeoffs between time resolution and accuracy have to be made. How best to make these tradeoffs is another important problem.

The signal to noise ratio is often very low and how to best handle noise is always an issue. Depending on the modeling assumptions made, the approach and complexity will vary. It is particularly challenging when a nonparametric approach is adopted since such a formulation allows for more general models for the current sources.

High resolution with limited measurements is always challenging and calls for more intricate signal processing techniques. Parametric models potentially provide a useful vehicle for extracting more information from the data. However, success of parametric model based schemes depend critical on their suitability for the problem. On the other hand, nonparametric methods allow for more general source models but high resolution estimates are harder to obtain. Currently for nonparametric methods, when data is limited, minimum norm solutions are computed and are popular. However, the criterion of minimum norm is not conducive to localized solutions. Devising approaches to reconstruct more general source models but yet impose some localization properties as required by the MEG problem is a challenging research task. It has more general implications and can be useful in many applications where nonparametric methods are used. In applications where localized solutions are desirable, combining minimum energy criterion with a minimum complexity criterion offers promise. In the MEG problem, solutions with minimal/reduced support, i.e. solutions with fewer nonzero entries, would be considered less complex. Attempts to define cost measures that formalize complexity, and developing efficient algorithms to minimize such cost functions hold great promise and pose an important challenge.

Work at UCSD

The main objective of our work has been to address the issues that arise in solving the MEG inverse/reconstruction problem. The main results of our research have been in the development of a novel nonparametric method called FOCUSS (FOCal Underdetermined System Solution) to solve the MEG reconstruction problem. An important attribute of the FOCUSS method is that it extracts compact but otherwise arbitrarily shaped areas of activation. Consequently, it has the flexible modeling capability of nonparametric methods and also provides high resolution estimates. An example of the capability of the algorithm is shown in figure 1. A 2-D geometry is used in these simulations. The planar geometry consisted of 17 sensors and a 72 point reconstruction grid. In the presentation format, sensors are indicated by circles and all source currents are assumed to be normal to the plane of reconstruction. As can be clearly seen, in contrast to the minimum norm method which does not yield localized reconstructions, FOCUSS yields high resolution reconstructions.

A complete analysis of the algorithm has also been conducted. In particular, global convergence of FOCUSS and local convergence rates of the algorithm have been established. Also procedures to deal with noise (Regularization in conjunction with FOCUSS) have been developed and studied. They show considerable promise. However, much remains to be done. Also extensive study of the FOCUSS algorithm for the

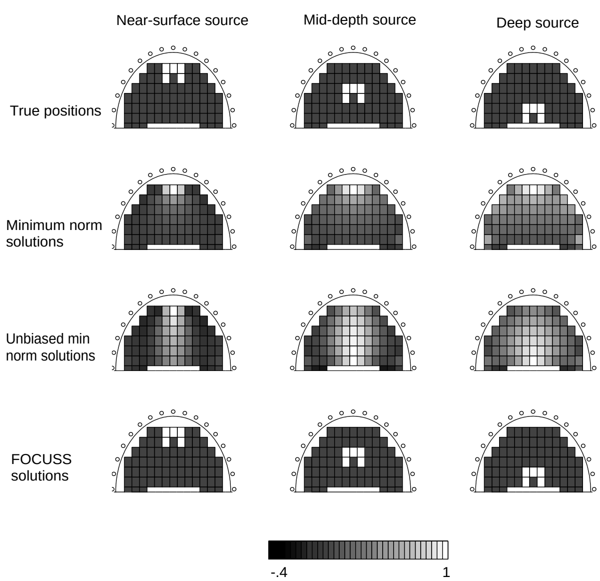


Figure 3.39: Resolution achieved with different reconstruction techniques for an extended source at different depths. Each column illustrates a different depth source (near-surface, mid- depth and deep) and each row illustrates a different reconstruction method (minimum norm, unbiased minimum norm, and FOCUSS). Simulated current distributions are illustrated in the top row. The compound version of FOCUSS with bias compensation was used.

MEG application has been conducted on synthetic data. They show the algorithm to be reliable.

Future Work

In our work on FOCUSS, we have taken some important steps to address the signal processing challenges that face MEG. However, much remains to be done to improve the reliability of the data processing in order for this useful technology to become clinically acceptable and economically viable. Issues relating to dealing with noise, multiple snapshots, resolution limits (both temporal and spatial), robustness to modeling uncertainty etc, still need further exploration before definitive conclusions can be drawn. Our future work consists of continuing to work on the MEG inverse problem and address the algorithmic challenges discussed above. In addition, there is a need to evaluate the methods developed on real data. This would involve using the algorithms in conjunction with anatomical information, and making realistic comparisons with other reconstruction methods.

3.7.2 Biomedical Applications of Array Signal Processing

Kevin Buckley, University of Minnesota

Applications, General Investigations and DSP Education

In this section, four research and lab development projects supported by the PYI Grant MIP-9057071 are described. The common research theme is Array Signal Processing. The purpose of the developed course and projects laboratory is real-signal and real-time Digital Signal Processing (DSP). The four supported projects are entitled:

- Basic Issues in Locating Radiating Sources,
- Microphone Arrays for Hearing Aids,
- Electroencephalogram (EEG) Array Processing, and
- DSP Education.

To date, this grant has supported five Ph.D. students in the department of Electrical Engineering at the University of Minnesota. Four of these students have already graduated.

Basic Issues in Estimating Locations of Radiating Sources

Estimating the locations of radiating sources, using signals from an array of sensors, is an important function both in traditional applications such as Communications, Geophysical Exploration, SONAR, RADAR and Astronomy, and in emerging array applications such as Hearing Aids and audio systems, Electroencephalogram (EEG) Studies and Crack Detection/Localization for Aircraft and Machinery. Certain performance issues are common to all applications. For the successful application of array processing to any particular problem, it is as important to understand these application independent issues as it is to have command of application specific challenges. Basically, if it is not understood how a location estimator performs under standard or idealistic situations, it is difficult to interpret results obtained from real-data applications. It is also difficult to select the best processing procedure for a given application without an understanding of basic characteristics of relative performance of a set of candidate procedures.

Contributions

Our research on basic source location estimation over the past several years has focused on performance analysis and on development of robust, highly accurate array signal processing algorithms. Within the context of this discussion, a highly accurate array signal processing algorithm is a computational procedure which provides, from a limited amount of array data, source location estimates with low bias and variance (i.e. estimates with small deviation from the actual location). A robust algorithm is a signal processing procedure which provides reliable estimates of source locations even when employed information about array, propagation channel and noise characteristics is inaccurate.

Concerning performance given limited data, we have developed an estimator variance and bias analysis approach and applied to several classes of highly accurate source location estimators. These include:

- the eigenspace based class (e.g. termed MUSIC, root-MUSIC, MinNorm, Closest, ESPRIT and Weighted Subspace Fitting), which have received considerable attention in the research literature in recent years;
- the optimum spatial filter based class (e.g. MVDR), which in application specific communities are often considered to be the most reliable; and
- variations of these two classes of methods which provide performance trade-offs.

Extending our results on limited data analysis, we have also been considering the important issue of sensitivity of high accuracy source location estimators to modeling errors, which result from inaccurate assumptions on source, noise, propagation channel and array characteristics.

The principal contribution of our effort is the ability to comprehensively compare a broad range of source location estimation algorithms which are of interest, including those listed above. The analytical measures we've developed for the estimators in issue are of:

- location estimate statistical variance,
- location estimator asymptotic bias, and
- location estimate statistical bias.

Besides comparing existing algorithms, we have used results from our analysis to identify new estimators which concurrently enhance high accuracy and robustness.

Our interest in robust source location estimators has lead us to a consideration the class of Bayesian based estimators of location parameters and numbers-of-sources. This Bayesian approach allows us to incorporate robustness into optimum estimators, by directly using statistical information about the uncertainty of assumed array, noise, propagation channel and source characteristics.

Our research has resulted in source-location and number-of-sources estimation procedures which provide improvement especially for limited data cases, including the critically important case where only one data sample per sensor is available.

Figures 3.40 and 3.41 exemplify results of this research. Figure 3.40 shows the relative performance of two popular source location estimators given uncertainty in the source observation model. MUSIC

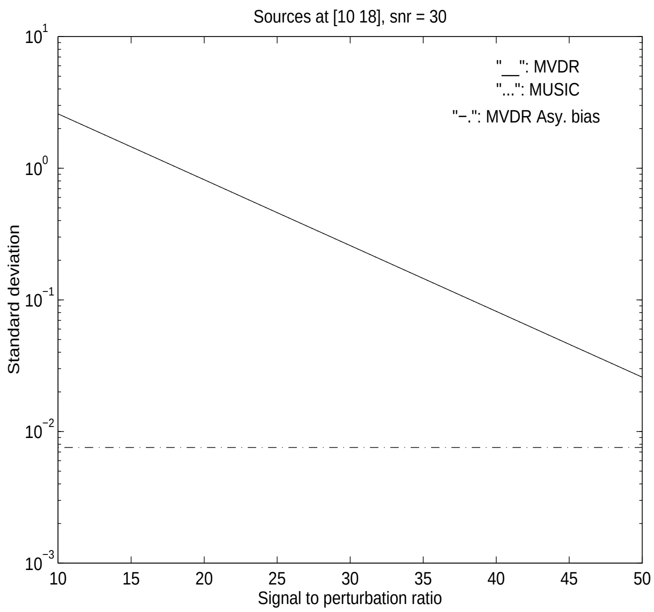


Figure 3.40: Comparison of two source location estimators.

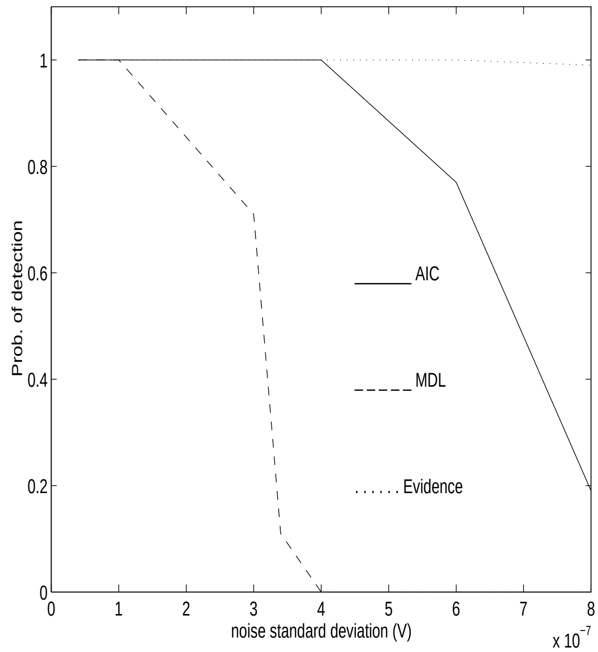


Figure 3.41: Comparison of number-of-source estimation methods.

(an eigenspace method) and MVDR (an optimum filter based method) are compared. Relative to MVDR, MUSIC is generally considered to be more accurate, but less robust and more computationally expensive. Figure 3.40 is a graph of estimator deviation from the true value vs. the amount of model uncertainty (left is more uncertain). We've shown that the primary reason MUSIC outperforms MVDR is that unlike MUSIC, MVDR is asymptotically biased (inaccurate even with infinite data). However, the graph shows that with even a small amount of modeling error the standard deviations of MUSIC and MVDR (which are virtually the same) dominate MVDR's asymptotic bias, and thus the two estimators will perform similarly. This suggests, and more importantly quantifies, the fact that if the more computationally expensive eigenspace procedure is to be used with advantage, we had better be very careful about designing and characterizing the data acquisition system.

Figure 3.41 investigates estimation of the number of sources. We compare a new Bayesian Evidence method with two popular methods (AIC and MDL). The graph shows the probability of detecting two dipole sources in a human brain (generated by cognitive activity) vs. the strength of background noise. Only a few data samples per EEG electrode were used. Our new method, which is specifically designed to provide accurate estimation when data is severely limited, clearly outperforms the standards.

Technical Challenges

As long as there is interest in considering the potential advantages of employing array data, either in new signal processing applications or to improve performance for existing applications, there will be a need for basic array processing research. High performance and robustness are generally conflicting objectives, since high performance required high sensitivity to characteristics of interest and robustness is defined as low sensitivity to characteristics not of interest. To jointly achieve both, flexible methods which can be finely tuned are needed, and specific applications must be well understood.

New applications and stricter performance requirements will continue to require both new array processing methods and new understandings of the performance of established methods. Continued investigation is required in the areas of:

- flexible array processing procedures and their performance, and
- application specific development.

Microphone Arrays for Hearing Aids

With the Audio DSP Laboratory and computational facility developed with PYI Grant matching funds, we have been conducting research into the application of multiple microphone arrays and digital signal processing to hearing aids. This research has focused on the enhancement of the intelligibility of a desired speech signal which is received in a reverberant (multipath) environment in competition with interfering speech and noise signals. We have developed spatial filters which attenuate reverberant and interference signals.

The current state-of-the-art hearing aid products are in-the-canal, analog signal-path devices. They provide the requisite frequency shaping and amplification for the user. However, they do not effectively reduce undesired noise. They therefore do not provide an increase in intelligibility of a desired signal which is considered needed for the hearing impaired in common noisy environments. With digital signal processing and multi-microphone hearing aids, there exists the potential of implementing a degree of flexibility and adaptability unrealizable with analog hardware. This can be used to provide optimum spatial filtering of noise in changing environments. To date there have been only a few digital signal processing hearing aids product development efforts. These have not targeted multimicrophone spatial filtering as a function, and have not resulted in profitable products. However, research into the use of multimicrophone hearing aids clearly indicates that significant improvement in intelligibility can be realized. We have identified an effective adaptive spatial filtering procedure for hearing aids, and evaluated its performance.

Contributions

The specific problem we targeted is noise reduction, through spatial filtering of data from multiple microphones, for hearing aids. As shown in Figure 3.42, the

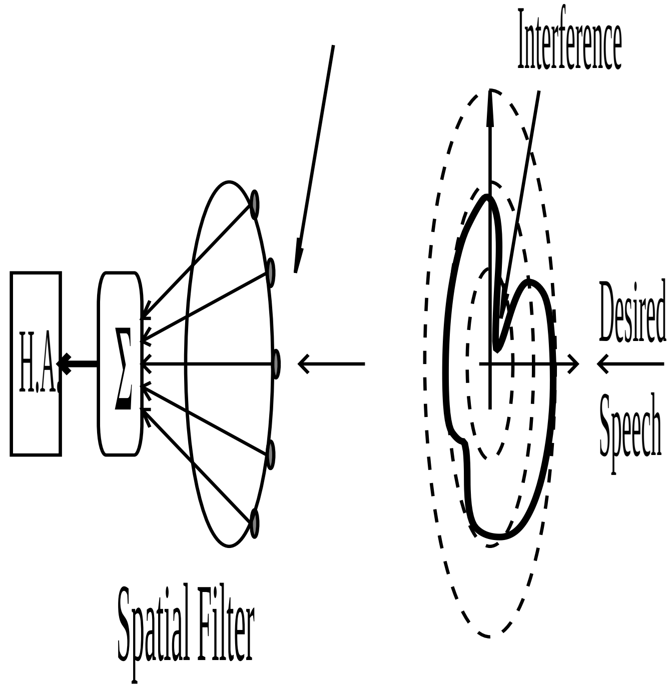


Figure 3.42: A head-worn spatial filter and a spatially selective response.

spatial filter is to be a preprocessor for a standard hearing aid which provides frequency shaping and amplification. The objective of the preprocessor is to attenuate interference and reverberation without appreciably distorting the desired signal. If the desired signal is distorted, the standard hearing aid will not provide the frequency shaping and amplification required for the specific user. The situation we've been considering is realistic and very challenging. The environment is reverberent and time varying. Noise sources are moving in space and turning on and off. The microphone array is broadband (speech spectral range), inexpensive (and therefore variable from device to device), and headworn (thus easily steered but very limited in aperture). The hearing aid preprocessor should be realized as a device which is no larger than a behind-the-ear aid. Fortunately, by usual array applications standards, only a modest increase in signal-to-noise ratio (SNR) is required to provide the hearing aid user with the increase in intelligibility needed to be at even performance level with a person of normal hearing.

Given these restrictions, and an understanding of their implications developed through realistic simula-

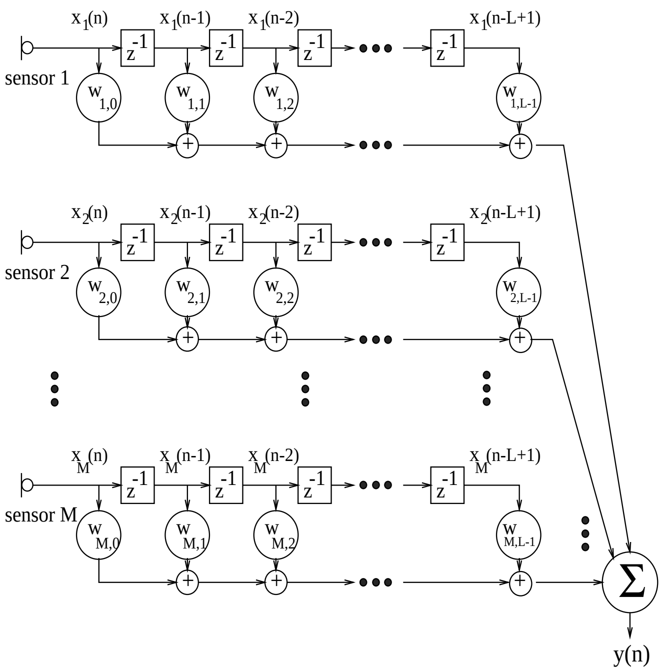


Figure 3.43: A multimicrophone spatial filter structure for hearing aids.

tions, we've identified the digital processing structure depicted in Figure 3.43 as being simple and adequate enough for this problem. Only three to seven microphones are required, and at a 10KHz sampling rate no more than 16 multipliers per microphone (represented in the figure as the $w_{i,j}$) are required. The key to effective spatial filtering is the selection of the $w_{i,j}$ multipliers. For this, we borrowed and modified as required techniques from basic sensor array processing. Since the noise and reverberation characteristics are different for different situations, and time varying, the weight selection algorithm adaptively minimizes output power. However, the set of multipliers are constrained such that the desired signal is processed without appreciable distortion, even though the head-worn array may vary in characteristics from user to user, and may not be steered directly at the desired speaker. Thus the process of output power minimization attenuates only reverberation and interference.

Table 1 shows intelligibility gain for the hearing aid spatial filtering approach we developed.

Reverberation and Case	Δ SNR	Δ SRT	G_I
Anechoic 7 mike	22.7	26.1	25.6
Anechoic 3 mike	15.4	18.8	18.0
Living Room 7 mike	10.2	10.4	10.6
Living Room 3 mike	5.1	7.3	6.1
Conf. Room 7 mike	3.1	3.6	3.8
Conf. Room 3 mike	1.4	2.2	1.7

Table 3.2: Intelligibility test results, Δ SNR, GI for several environments and processors

It shows results of formal subject tests. Ten normal hearing subjects were asked to identify spondaic words in reverberation and noise. A realistic simulation program was used to generate both sensor output and spatial filter output signals. This simulation program incorporated reverberation, microphone placement errors (on a head), acoustical effects of the head, and inaccuracy in steering the headworn array towards the desired speaker. An interferer was simulated at 45 degrees to the right of the desired speaker. Results of six cases are shown, where the amount of reverberation (none = anechoic, modest = living room, significant = conference room) and number of microphones were varied. The third column, labeled ΔSRT, shows the results of the formal listening tests. Note that with seven microphones, an improvement in speech reception threshold (SRT) of almost 4 dB is realized. This means that, compared to the output of the best microphone (that with the highest intelligibility), by using a seven microphone spatial filter we have effectively increased the desired signal power by almost 4 dB. Columns 2 and 4, labeled ΔSNR and G_I respectively, show the calculated improvement in signal-to-noise ratio and calculated improvement in intelligibility. Two results are worth noting. First, we realize the required improvement, even in a very challenging reverberent environment. Second, G_I, a measure we derived for calculating intelligibility improvement from microphone and spatial filter output signals, accurately reflects intelligibility improvement. G_I can thus be used to evaluate hearing aid performance, without having to resort to expensive and time consuming formal listening tests. Consequently we have used this measure to verify that the proposed spatial filtering provides more than the targeted improvement in many realistic situations.

We have also developed real-time implementations of these hearing aid spatial filtering algorithms, using the Motorola DSP56001 Digital Signal Processing chip. Besides the primary task of implementation and evaluation for the hearing-aid application, this involved the development of a new hearing-aid array calibration method, which is simple and effective enough to be potentially implemented in a hearing-aid fitting situation.

Technical Challenges

It has been established that adaptive microphone array spatial filters can be used to significantly reduce noise levels (relative to single microphone and fixed multimicrophone spatial filters) in realistic hearing aid situations, where there is reverberation, noise, substantial uncertainty in signal observation models, and limited array and processor resources. The challenge now is to convince the hearing aid manufacturing and consumer communities of this advantage, and to realize the advantage with competitive products.

Electroencephalogram (EEG) Array Processing

Figure 3.44 illustrates the EEG electrode configuration commonly used today. Electrode signals are studied in both clinical and research situations, for example, to investigate sleep disorders, to localize sources of epileptic seizures, to detect AIDS dementia and to study brain functionality. Individual electrode outputs are evaluated visually or with computer assistance, to study temporal and frequency characteristics. Spatial origin of signals, within the brain, is qualitatively determined by inspecting several electrode outputs at the same time.

Formal EEG electrode array processing is being investigated. We have been actively researching one objective – the accurate localization of sources of cognitive activity.

Contributions

Our investigation into EEG array processing has been concerned with:

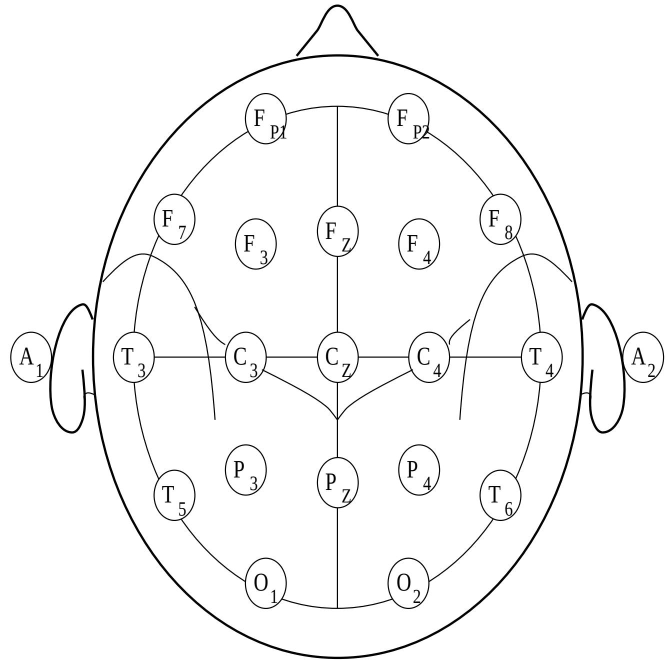


Figure 3.44: The standard 10/20 EEG electrode configuration.

- EEG electrode configuration design, for which we have developed a theoretically based sampling theorem and a practical method of determining the required number of EEG electrodes for a given task;
- robust dipole source localization and number-of-source determination, for which we have successfully applied the general Bayesian approach described earlier; and
- the application and evaluation of spectral based dipole localization methods, for which we are extending the performance analysis discussed above to the multidimensional source location parameter problem.

Figure 3.41 illustrates one of our contributions to EEG array processing. Two dipole sources were simulated in the cortical region of the brain. Given were 35 EEG electrodes and 50 data points per electrode. As noted earlier, the graph compares several methods for estimating the number of sources, and clearly demonstrates the advantage of our new Bayesian Evidence method for situations where there is limited data.

Technical Challenges

Although EEG has been around for a long time, it usefulness as an approach to spatial mapping of cognitive cortical activity is still in question. Further research on this topic is warranted because EEG (along with Magnetoencephalogram (MEG)) Array Processing offers two things that other brain imaging modalities can not:

- a direct measure cortical electromagnetic activity; and
- millisecond temporal resolution.

Additionally, EEG is by far the least expensive brain imaging modality.

Comprehensive, multidisciplinary investigations are needed. Application specific array processing algorithms need to be developed using best available general array processing methods. For example, robust Bayesian parameter and number-of-sources estimation methods need to be considered in addition to the standard least-squares and spatial-spectral methods that have been suggested. Algorithm development is needed based on both detailed simulations and real data which is acquired under careful control. Currently, the biggest impedement to the development of effective EEG array processing algorithms is lack of reliable data. There are indication that this problem will be resolved in the near future.

DSP Education

With donated equipment from Motorola, University of Minnesota financial and facility support, and matching funds from NSF, a Digital Signal Processing (DSP) Laboratory has been developed which is currently used for:

- an undergraduate general DSP Lab Course (approximately 70 students/year);
- a graduate real-time DSP Lab Course (approximately 15 students/year); and
- DSP Projects and Research.

Figure 3.45 illustrates this lab.

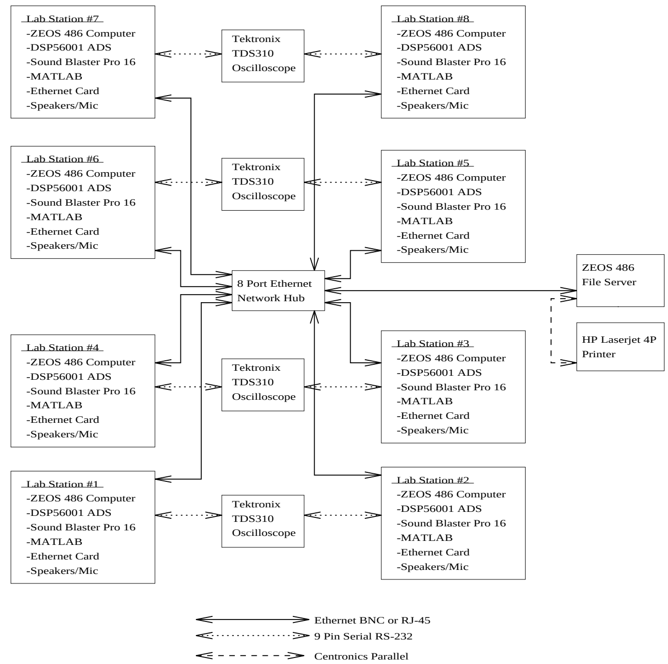


Figure 3.45: The DSP Laboratory.

The Lab Courses provide students hands-on experience with real-data and real-time signal processing, an experience consistently sought by local industry. Students investigate methods for: spectrum estimation; digital filter design and implementation; real-time adaptive and multirate filtering; and full-duplex, real-time differential pulse code modulation. Lab Projects to date include: real-time noise cancellation for EKG signals; real-time spatial filtering for hearing aids and other audio applications; real-time active acoustic noise cancellation; and material crack detection, localization and diagnostics based on acoustic emissions generated during crack formation.

3.7.3 SAR Image Formation and Processing

Jian Li, University of Florida

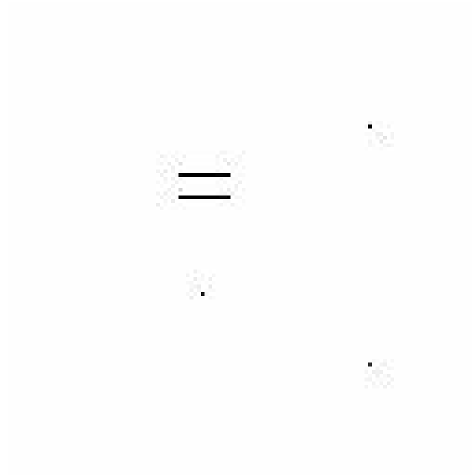
SAR Image Formation and Processing

The objective of our research efforts on synthetic aperture radar (SAR) image formation and processing is to enhance SAR image formation and understanding techniques. Our efforts have potential impact on both environmental monitoring and military and law enforcement applications. For environmental monitoring, our results can be used to better detect, analyze, and quantify environmental changes. In military and law enforcement applications, our results can be used to better extract target features for target recognition.

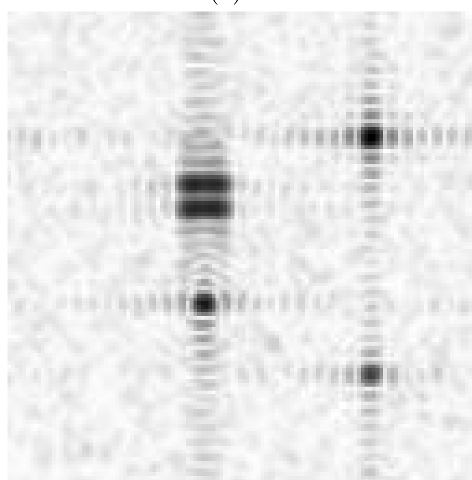
We have so far primarily focused on SAR image formation via spectral estimation methods. We have found that existing parametric methods are not applicable in general to the SAR image formation problem. We have investigated how nonparametric methods such as Capon and Welch methods, which are FIR (finite impulse response) filtering approaches, can be used for SAR image formation. We have also developed an adaptive FIR filtering approach method, which is referred to as the APES algorithm, for spectral estimation and SAR imaging. We have shown via both numerical and experimental examples that the adaptive FIR filtering approaches such as Capon and APES can yield more accurate spectral estimates with much lower sidelobes and narrower spectral peaks than the FFT method, which is also a special case of the FIR filtering approaches. We show that although the APES algorithm yields somewhat wider spectral peaks than the Capon method, the former gives more accurate overall spectral estimates and SAR images than the latter and the FFT method.

Figure 3.46(a) shows the modulus of a simulated SAR image. Note that the data consists of 3 spectral lines and two closely-spaced one-dimensional continuous pulses. (The line spectra in Figure 3.46(a) simulate corner reflectors and the 1-D continuous pulses simulate dihedrals in SAR images.) Figure 3.46(b) shows the modulus of the spectral estimate obtained with the 2-D FFT method. The use of the 2-D FFT results in high sidelobes; the two 1-D continuous pulses in the spectrum are barely resolved. Figure 3.46(c) shows the modulus of the spectral estimate obtained by using 2-D FFT with a circularly symmetric Kaiser window. The windowed FFT method reduces the sidelobes. However, it widens the spectral peaks and, as a result, the closely spaced one-dimensional continuous pulses are smeared together. Figures 3.46(d) and (e) show the spectral estimates obtained by using the 2-D Capon and 2-D APES, respectively. Note that 2D-APES gives slightly wider spectral peaks than 2D-Capon, but the spectral estimates obtained with the former are more accurate.

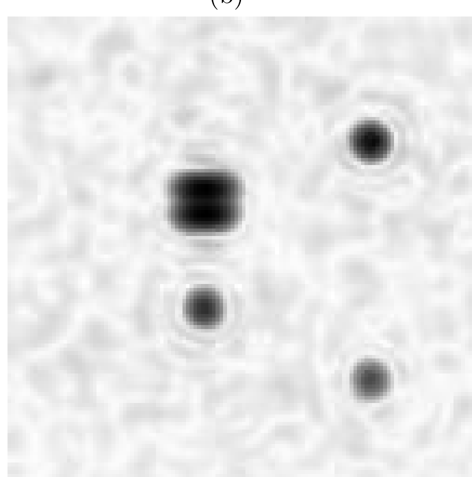
Figure 3.47 shows an example of the results obtained with the experimental data collected by ERIM (Environmental Research Institute of Michigan). Figures 3.47(a) and (b) show the SAR images obtained with the 2-D FFT method and the 2-D APES method. Note that the SAR image formed with the latter method has



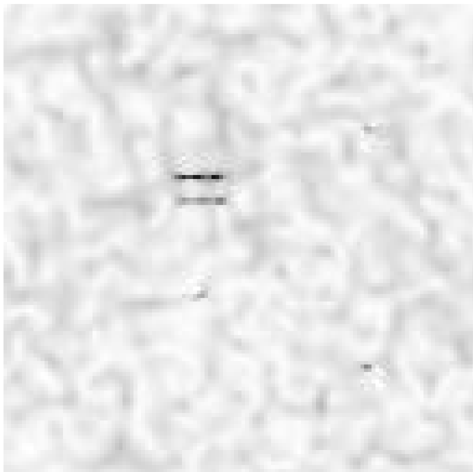
(a)



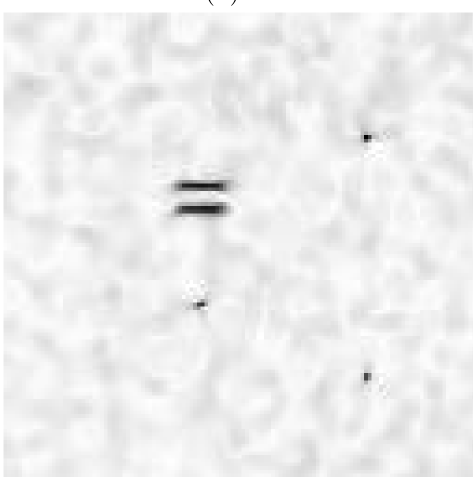
(b)



(c)

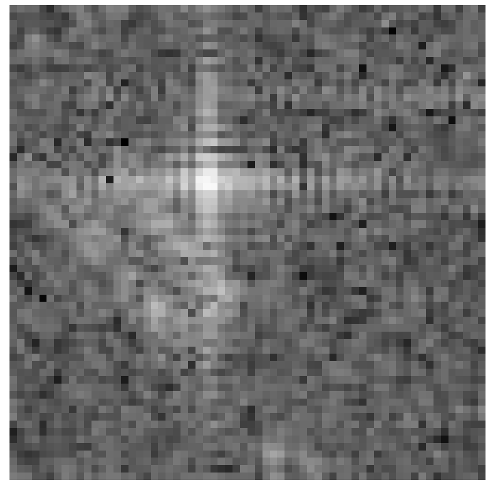


(d)

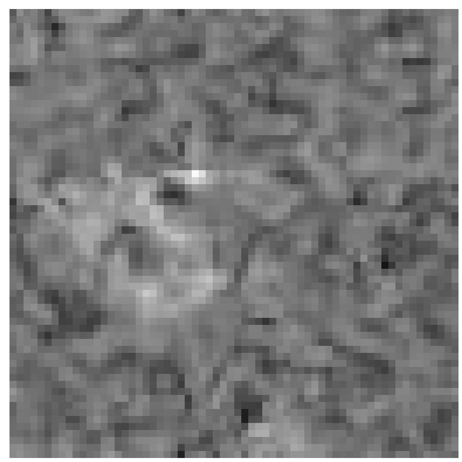


(e)

Figure 3.46: SAR image estimates. (a) True SAR image. (b) 2-D FFT. (c) 2-D FFT with circularly symmetric Kaiser window. (d) 2-D Capon.(e) 2-D APES.



(a)



(b)

Figure 3.47: SAR image estimates. (a) 2-D FFT. (b) 2-D APES.

much lower sidelobes and less speckle than the one obtained with the former method.

We have also made initial progress on improving the convergence rate of the K-means and adaptive K-means algorithms used for image segmentation and understanding. Fast convergence is needed when processing a large amount data used in environmental monitoring. For the K-means algorithm, the means are updated locally rather than globally. For the adaptive K-means algorithm, we improve its convergence rate via hierarchical implementation and multiresolution wavelet decomposition.

Future Direction

To devise practical array processing algorithms, we must emphasize on the robustness of the algorithms, which must be based on realistic data models. Our future research direction is to establish realistic data models and devise robust algorithms to solve practical problems in the many application areas we are interested in including communications and radar.

3.7.4 Null Steering/ Beamforming Arrays Used in Conjunction With GPS Receivers

Anton S. Gecan, E-Systems and Michael D. Zoltowski, Purdue University

Objectives

The goal of this project is to develop high-speed adaptive null steering array systems for the purpose of protecting the GPS signal from interference. Recent investigations by both the Defense Science Board and a subcommittee of the National Research Council have led to the conclusion that the GPS signal is quite vulnerable to interference - either deliberate or inadvertent.

Progress to Date and Milestones

The approach taken is to find that set of spatial filter weights that minimizes the output power of the beamformer subject to a single linear constraint on the weights: unity weight on the reference element (*e.g.,* the center element of a circular array.) This approach leads to a computationally simple algorithm that is robust to calibration errors due to mismatch amongst the antenna elements comprising the array.

The power minimization approach is premised on the fact that the GPS signals are below the noise floor and that the respective signals from different GPS satellites may be selected based on their corresponding PN sequences *after* beamforming, and may thus pass through the beamformer simultaneously. The algorithm works to drive the output power down to the noise floor thereby putting nulls in the directions of the interfering sources.

The conjugate gradient method is used to minimize the output power subject to a unity weight constraint on the reference element. Through a simple change of coordinate bases, the constrained optimization problem may be converted to an unconstrained one. A simulation is presented to demonstrate the effectiveness of this approach at suppressing interferers under stressed conditions.

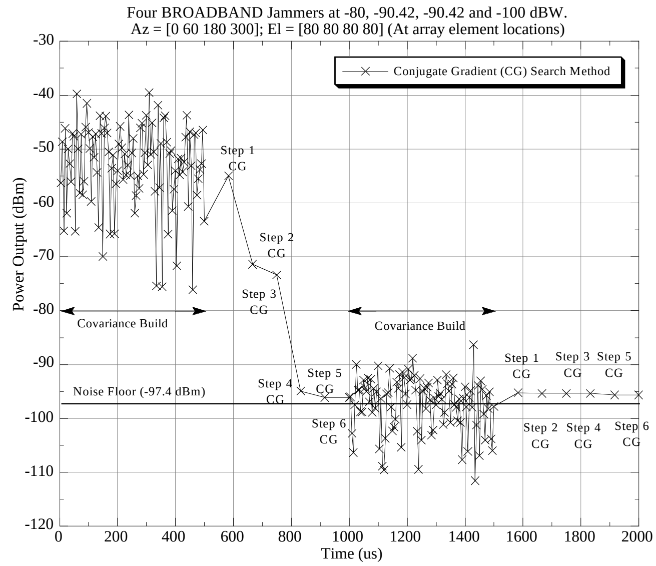
The AE1-α prototype antenna array built by E-Systems consists of six antennas equi-spaced around a circle of radius 3.75 inches and an additional antenna at the center of the circle serving as a reference element. The jammer scenario involved 4 broadband Gaussian noise sources equi-spaced in azimuth at 10° elevation measured up from the horizon (80° with respect to boresite.) The powers of the jammers ranged from -80 dBW down to -100 dBW. The 7×7 spatial covariance matrix is built up during the first 500 microseconds (μs) from a total of 100 snapshots, one every 5 μs. The instantaneous output power obtained at each snapshot during this covariance build-up period with unity weight on the reference element and zero weight on all of the auxiliary elements is indicated by the first 500 μs segment of the power versus time curve plotted in Figure 3.48 (a).

During the next 500 μs, a conjugate gradient (CG) search is run. In the case of strong interferers, the CG search requires a maximum of 6 steps to converge to the minimum possible output power under the unity weight constraint on the reference element. Each step requires 80 μs of computation time. Although it is an intermediate quantity, the average output power obtained with the set of weights at each of the 6 iterations of the CG search is plotted as that 500 μs segment of the power versus time curve in Figure 3.48 (a) running from 500 μs to 1000 μs. This is an average power measurement based on the time-averaged covariance matrix. It is observed that convergence occurs in four steps of the CG search, which is equal to the number of jammers in this case. In addition, for this set of jammer locations and corresponding powers, the final average output power is 4 dB above the noise floor.

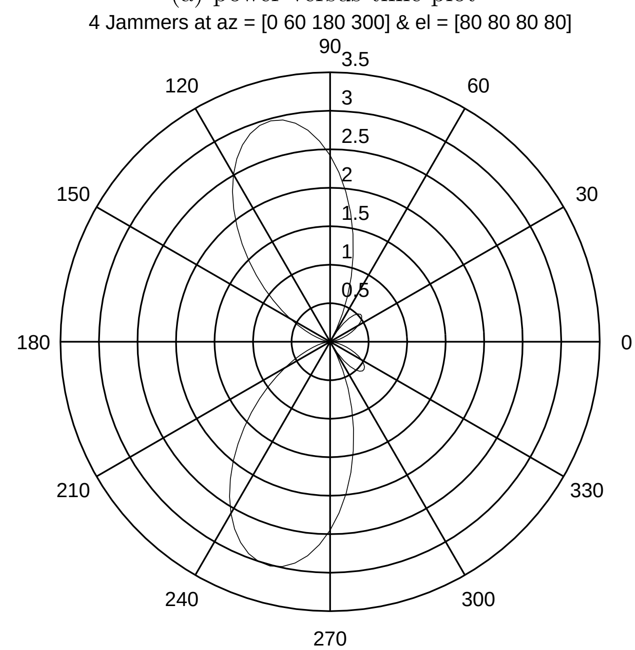
Note that during the 500 μs interval the CG search is being run, another processor is simultaneously building an update of the spatial covariance matrix. The instantaneous power obtained at each snapshot during the 500-1000 μs time interval with the initial set of weights is not shown in Figure 3.48.

Since the final output power obtained from the CG search at 1000 μs is an average power based on a time-averaged covariance matrix, during the 1000-1500 μs time interval we plot the instantaneous power obtained with the optimum set of weights output from the CG search; this reflects noise and some residual jammer signals. Keep in mind that during this 1000-1500 μs time interval two processes are occurring simultaneously: (1) a new covariance matrix is being formed and (2) the conjugate gradient search is being run with the covariance matrix formed during the 500-1000 μs time interval.

Figure 3.48 (b) shows the azimuthal beam pattern at the common elevation of the interferers obtained with the set of weights output from the CG search at 1000 μs; nulls are observed at the respective azimuth angles of the four jammers.



(a) power versus time plot



(b) azimuthal beam pattern at jammer elevation

Figure 3.48: Simulation of power minimization via conjugate gradient method with full covriance build after complex baseband demodulation and A/D conversion: four broadband jammers with diverse powers at common low elevation angle of 80°.

Appendix A

Participants: Addresses

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Appendix B

Participants: Biographies

Rick S. Blum received a B.S. in Electrical Engineering from the Pennsylvania State University in 1984 and his M.S. and Ph.D in Electrical Engineering from the University of Pennsylvania in 1987 and 1991. From 1984 to 1991 he was a member of technical staff at General Electric Aerospace in Valley Forge, Pennsylvania and he graduated from GE's Advanced Course in Engineering. Since 1991, he has been an Assistant Professor in the Electrical Engineering and Computer Science Department at Lehigh University in Bethlehem, Pennsylvania. His research interests include signal detection and estimation and related topics in the areas of signal processing and communications. He is a senior member of IEEE and a member of Eta Kappa Nu and Sigma Xi. He holds a patent for a parallel signal and image processor architecture.

Yoram Bresler received the B.Sc. (cum laude) and M.Sc. degrees from the Technion, Israel Institute of Technology, in 1974 and 1981 respectively, and the Ph.D degree from Stanford University, in 1985, all in Electrical Engineering. From 1974 to 1979 he served as an electronics engineer in the Israeli Defense Force. From 1979 to 1981 he was a consultant for the Flight Control Lab at the Technion, Israel, developing algorithms for autonomous TV aircraft guidance. From 1985 to 1987 he was a Research Associate at the Information Systems Laboratory at Stanford University, where his research involved sensor array processing and medical imaging. In 1987 he joined the University of Illinois at Urbana-Champaign, where he is currently an Associate Professor at the Department of Electrical and Computer Engineering and the Bioengineering Program, and Research Associate Professor at the Computer and Systems research Lab. His current research interests include multidimensional and statistical signal processing and their applications to inverse problems in imaging and sensor array signal processing, and to diagnostic and scientific visualization. Dr. Bresler was an Associate Editor for the *IEEE Transactions for Image Processing* in 1992-93 , and currently is on the editorial board of *Machine Vision and Applications,*, and a member of the IEEE Image and Multidimensional Signal Processing Technical Committee. In 1988 and 1989 he received the Senior Paper Awards from the IEEE Acoustics, Speech, and Signal Processing society. He is the recipient of a 1981 Fulbright grant for doctoral studies, and a 1991 Presidential Young Investigator Award of the National Science Foundation.

Kevin M. Buckley was born in Washington D.C., Dec. 1954. He received B.S. and M.S. degrees in Electrical Engineering at Villanova University in Pennsylvania in 1976 and 1980, respectively, and the Ph.D. degree from the University of Southern California in 1986. From 1980 to 1982, Dr. Buckley worked at Sonic Sciences Inc., Warminister Pa., on communication and sonar applications of digital signal processing. In 1982-83, he worked as a full time instructor at Villanova University. Since 1986 he has been on the faculty of the Department of Electrical Engineering, University of Minnesota, where he is currently an associate professor. Dr. Buckley has been the chair of the SP Society Technical Committee on Statistical Signal and Array Processing from 1992 to 1994, and is currently a member of that committee. He was an Associate Editor for the IEEE Transactions on Signal Processing from 1992 to 1994, he has served as Technical Committee Co-chair for ICASSP-93 and Co-chair for the Fourth IEEE Workshop on Spectrum Estimation and Modeling. Dr. Buckley received the 1989 Gallen (Villanova University Distinguished Engineering Alumnus) Award, and has been selected as the recipient a 1990 National Science Foundation Presidential Young Investigator Award. His primary research and teaching interests involve the theory and application of: sensor array processing, parameter estimation, adaptive filtering and general signals and systems.

Ronald D. DeGroat was born in Lyons, New York on April 12, 1957. He received his B.S in Physics from Baylor University in 1979, his M.S. in Physics from the University of Texas at Dallas in 1981, and his Ph.D. in Electrical Engineering from the University of Colorado at Boulder in 1987. In 1987, he joined the faculty in Electrical Engineering at the University of Texas at Dallas where he is currently an Associate Professor. His main research interests include eigenstructure and subspace-based signal processing, fast algorithms, array processing methods and nonlinear systems modelling.

Eric M. Dowling was born in Oak Park Illinois on April 9, 1962. He received the B.S.E.E, M.S.E.E. and Ph.D. degrees from the University of Florida in 1984, 1986 and 1989 respectively. While at the University of Florida, he consulted for the Athena Group Inc. and served in the Florida Army National Guard as a multichannel systems platoon leader, battalion radio officer, and corps area signal company commander. In 1989 he joined the electrical engineering faculty of the University of Texas at Dallas as an assistant professor. His research interests include adaptive and subspace based signal processing, numerical linear algebra, optical lattice filters, and systolic architectures. He consults and performs research with industry and government regularly.

James Flanagan is Vice President for Research at Rutgers University. He is also Board of Governors Professor in Electrical and Computer Engineering, and Director of the CAIP Center. Flanagan joined Rutgers after extended service in research and research management at AT&T Bell Laboratories. He was previously Director of Information Principles Research, with responsibilities in digital communications and information systems. Flanagan holds the S.M. and Sc.D. degrees in Electrical Engineering from the Massachusetts Institute of Technology. He has specialized in voice communications, computer techniques and electroacoustic systems, and has authored approximately 150 papers, 2 books, and 45 patents in these fields. Flanagan is a Fellow of the IEEE, the Acoustical Society of America, and the American Academy of Arts and Sciences. He has received a number of technical awards, and is a member of the National Academy of Engineering and of the National Academy of Sciences.

Anton Gecan received his B.S.E.E. from Purdue University in 1971, an M.S. in Operations Research at Northeastern University in 1979, and an M.S.E.E. and Ph.D. at the University of South Florida in 1984 and 1992, respectively. Dr. Gecan is presently a Staff Engineer at E-Systems (ECI Division) in St. Petersburg, FL, where he is leading algorithmic development efforts on phased array and null steering antennas. Dr. Gecan is an Adjunct Professor at Florida Institute of Technology and the University of South Florida where he has taught courses in Computer Simulation, Statistics, and Telecommunications. Dr. Gecan served as a Navy Officer (Destroyers) and is a Vietnam Veteran.

Saleem A. Kassam received the B.S. degree in Engineering from Swarthmore College, Swarthmore, PA, in 1972, the M.S.E. and M.A. degrees in Electrical Engineering from Princeton University, Princeton, N.J., in 1974, and the Ph.D. degree in Electrical Engineering from Princeton in 1975. Since 1975 he has been on the faculty of the Moore School of Electrical Engineering, University of Pennsylvania, Philadelphia, PA, where he is currently the Solomon and Sylvia Charp Professor of Electrical Engineering. He served as Chairman of the Department of Electrical Engineering in the Moore School from 1992 to 1994. His research interests are in the area of statistical signal processing and communications, and he has made contributions to signal detection theory, quantization, robust signal processing, nonlinear and adaptive filtering, image processing, spectrum estimation, and microwave and acoustic imaging. He has co-edited a book on nonparametric detection, and is the author of "Signal Detection in Non-Gaussian Noise" (Springer-Verlag, 1988). He holds two patents on array imaging systems. Dr. Kassam was an Associate Editor for the IEEE Transactions on Information Theory (1990-1992) and is a member of the Board of Governors of the Information Theory Society. He is a member of Phi Beta Kappa and Sigma Xi, and is a Fellow of the Institute of Electrical and Electronics Engineers.

M. Kaveh received the B.S. and PhD degrees from Purdue in 1969 and 1974 respectively and the M.S. degree from the University of California at Berkeley in 1970. He has been at the University of Minnesota since 1975, where he is presently a Professor and the Head of the Department of Electrical Engineering. He was a design engineer at Scala Radio Corporation in 1970 and has consulted for industry including the MIT Lincoln Laboratory, 3M and Honeywell. He is a Fellow of IEEE and serves the IEEE Signal Processing Society as its Vice President for Publications. He is the recipient (with A. Barabell) of a 1986 ASSP Senior Award and the 1988 ASSP Meritorious Service Award.

Jian Li received the M.Sc. and Ph.D. degrees in electrical engineering from The Ohio State University, Columbus, in 1987 and 1991, respectively. From April 1991 to June 1991, she was an Adjunct Assistant Professor with the Department of Electrical Engineering, The Ohio State University, Columbus. From July 1991 to June 1993, she was an Assistant Professor with the Department of Electrical Engineering, University of Kentucky, Lexington. Since August 1993, she has been an Assistant Professor with the Department of Electrical Engineering, University of Florida, Gainesville. Her current research interests include sensor array signal processing, synthetic aperture radar image formation and understanding, radar detection and estimation theory, image segmentation and processing, and communications.

K.J. Ray Liu received the Ph.D. degree in Electrical Engineering from the University of California, Los Angeles, in 1990. Since then he has been a faculty member of Electrical Engineering Department and Institute for Systems Research, University of Maryland, College Park. Dr. Liu is the Director of Digital Signal Processing Laboratory and Co-Director of VLSI Design Automation Laboratory at the University of Maryland. His research interests span all the aspects of high performance computational signal processing including parallel and distributed processing, fast algorithm, VLSI, and concurrent architecture, with application to image/video, radar/sonar, communications, and medical and biomedical technology. Dr. Liu was the recipient of the National Science Foundation Young Investigator Award in 1994. He received the IEEE Signal Processing Society's 1993 Senior Award and was awarded the George Corcoran Award for outstanding contributions to electrical engineering education at the University of Maryland. Dr. Liu is an Associate Editor of IEEE Transactions on Signal Processing and is a member of VLSI Signal Processing Technical Committee of the IEEE Signal Processing Society.

Bhaskar D. Rao received the B. Tech. degree in electronics and electrical communication engineering from the Indian Institute of Technology, Kharagpur, India in 1979, and the M.S. and Ph. D degrees from the University of Southern California, Los Angeles in 1981 and 1983, respectively. Since 1983, he has been with the University of California, San Diego, where he is currently a Professor in the Electrical and Computer Engineering Department. His interests are in the areas of Digital Signal Processing, Estimation Theory and Optimization Theory with applications to communications, sonar, biomedical imaging, and speech.

Richard Roy III received his B.S. EE and B.S. Physics degrees from M.I.T. in 1972, an M.S. Physics degree from Stanford in 1973, and the M.S. EE and Ph.D. EE degrees from Stanford in 1975 and 1987 respectively. His Ph.D. research was in the field of digital signal processing, specifically estimation, parameter identification. Dr. Roy has been employed by ESL as a senior member of the technical staff involved in research and development of state-of-the-art techniques of information processing applied to intelligence data analysis, by MacLeod Laboratories, involved in research into and development of algorithms for real-time signal processing and information extraction from data collected in the oil-well drilling environment, and by Integrated Systems where he was involved in the development of state-of-the art techniques in estimation, identification, and adaptive control for various aerospace applications. During this period, he was also a member of the Information Systems Laboratory at Stanford University and a consultant to industry working in the areas of parameter estimation and system identification with applications to control and communications systems. In April of 1992, he co-founded with Marty Cooper, formerly with Motorola, ArrayComm, Inc., a California corporation involved in applying intelligent antenna technology to wireless telecommunications, and is currently chief technical officer. Dr. Roy is widely published internationally and holds several patents in the area of intelligent antenna technology and wireless communications.

Harvey Silverman is the Dean of Engineering at Brown University. He is also Professor of Engineering and the Director of the Laboratory for Engineering Man/Machine Systems at Brown. His current research interests include speech recognition and digital signal processing, parallel and reconfigurable computer architectures and all aspects of microphone-array work. Dr. Silverman was named Dean of Engineering at Brown after serving in various capacities at Brown. He had been named Professor of Engineering at Brown in 1980, after having spent 10 years at IBM Thomas J. Watson Research Center in Yorktown Heights, NY. At IBM, Dr. Silverman worked on image processing, computer-performance analysis, and algorithms and systems for speech recognition. He was Manager of a hardware speech recognition effort throughout the latter half of the 1970's. He holds a PhD and ScM degrees from Brown University and undergraduate degrees from Trinity College in Hartford, CT. He has published about 150 papers to date. Dr. Silverman was the Chairman, Technical Committee on Digital Signal Processing for the ASSP Society of IEEE, and the General Chairman, IEEE International Conference on ASSP in 1977. He was awarded an IEEE Centennial Medal in 1984 and has several invention and patent awards from IBM. He was appointed a Trustee of Trinity College, Hartford, CT in 1994.

Jon Sjogren has been Program Manager of the research activity in Signal Processing, Communications and Statistics at the Air Force Office of Scientific Research (AFOSR), located at Bolling Air Force Base, Washington DC, for the past five years. He joined the government in 1986 and contributed to Army and NASA projects in fault-tolerant computing for flight control, and formal proof-of-correctness methods for software, at the Langley Research Center (Hampton, VA). Sjogren received the Ph.D. degree from the University of California, Berkeley in 1975 (in mathematics) and is known for work in commutativity properties of groups. He has taught in France, Germany, the Middle East, Oregon and North Carolina. His current interests include the use of algebraic methods in multi-rate filtering (wavelets), neural networks and communications technology. He sees a clear role for the mission-directed research funding agencies in forming cohesive, multi-disciplinary groups of investigators, focusing attention on a class of problem areas for rapid progress in the interests of an explicit customer base.

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